



# **SRG50 Configuration Guide**

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## **BCM50 3.0** Survivable Remote Gateway

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# Chapter 1

## Getting started

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### About this guide

The *SRG50 Configuration Guide* describes how to install, configure, and maintain the Survivable Remote Gateway (SRG) 50 Release 3.0.

The SRG50 is positioned as a cost-effective Small IP Branch Office solution for CS 1000 Main office systems. The SRG50 offers business continuity and public switched telephone network (PSTN) failover for voice over IP (VoIP) networks. An SRG provides transparent operation, feature and application parity with a main office call server while in normal operating mode. If connectivity with the call server or wide area network (WAN) is lost, the normal mode sets revert back to local mode. The SRG takes ownership of call control for the local sets automatically and provides internal communications as well as external connectivity to the PSTN.

SRG50 supports H323 and SIP Trunking and up to 80 survivable IP users with a single SRG application authorization code. It is provided as a cost-effective VoIP business continuity solution for small branch offices. The SRG50 supports CS 1000 Release 4.0, 4.5, 5.0, and 5.5.

### Audience

The *SRG50 Configuration Guide* is intended for two audiences:

- the individuals responsible for engineering the SRG50 site and installing the BCM50, configuring it for operation as an SRG50, and connecting it to the network
- the individuals responsible for post-installation system administration and maintenance.

The SRG50 site engineer and installer must be familiar with BCM50 hardware and software, and IP telephony and VoIP trunk configuration on the BCM50.

### Acronyms

The following is a list of acronyms used in this guide.

**Table 1** Acronyms used in this guide (Sheet 1 of 2)

Acronym	Description
ACR	Alternative call routing
ANBWM	Adaptive network bandwidth management
ASM	Analog station module
ATA	Analog terminal adapter

**Table 1** Acronyms used in this guide (Sheet 2 of 2)

<b>Acronym</b>	<b>Description</b>
BARS	Basic alternate route selection
BUID	Branch user ID
CDP	Coordinated dialing plan
DDIM	Digital drop and insert mux
DN	Directory number
DSC	Distant steering codes
DSM	Digital station module
DTM	Digital trunk module
ESA	Emergency services access
ESDN	Emergency services DN
FRL	Facility restriction level
GATM	Global analog trunk module
KEM	Key expansion module
KRS	Keycode retrieval system
LAN	Local area network
LSC	Local steering codes
MCDN	Meridian customer defined network
MOTN	Main office terminal number
MVC	Mobile voice client
NARS	Network alternate route selection
NBWM	Network bandwidth management
NCS	Network connection server
NRS	Network routing service
PSTN	Public switched telephone network
QoS	Quality of service
SPN	Special number
SRG	Survivable remote gateway
TAT	Trunk anti-tromboning
TRO	Trunk route optimization
TSC	Trunk steering codes
UDP	Uniform dialing plan
VNR	Vacant number routing
VoIP	Voice over internet protocol
VPN	Virtual private network
VPNI	Virtual private network ID
WAN	Wide area network
ZDP	Zone digit prefix

## Symbols and conventions used in this guide

These symbols highlight critical information for the SRG system.



**Caution:** Alerts you to conditions where you can damage the equipment.



**Danger:** Alerts you to conditions where you can get an electrical shock.



**Warning:** Alerts you to conditions where you can cause the system to fail or work improperly.



**Note:** Alerts you to important information.



**Tip:** Alerts you to additional information that can help you perform a task.



**Security Note:** Indicates a point of system security where you can change a default or where the administrator must decide on the level of security required for the system.



**Warning:** Alerts you to ground yourself with an antistatic grounding strap before performing the maintenance procedure.



**Warning:** Alerts you to remove the main unit and expansion unit power cords from the AC outlet before performing any maintenance procedure.

These conventions and symbols represent the Business Series Terminal display and dialpad.

Convention	Example	Used for
Word in a special font (shown in the top line of the display)	P <code>swd</code> :	Command line prompts on display telephones.

Convention	Example	Used for
Underlined word in capital letters (shown in the bottom line of a two-line display telephone)	<u>PLAY</u>	Display options on two-line display telephones. Press the button directly below the option on the display to proceed.
Dialpad buttons	#	Buttons you press on the dialpad to select a particular option.

These text conventions are used in this guide to indicate the information described.

Convention	Description
<b>bold Courier text</b>	Indicates command names, options, and text that you must enter. Example: Use the <b>info</b> command. Example: Enter <b>show ip {alerts   routes}</b> .
<i>italic text</i>	Indicates book titles.
plain Courier text	Indicates command syntax and system output (for example, prompts and system messages). Example: Set Trap Monitor Filters
<b>FEATURE HOLD RELEASE</b>	Indicates that you press the button with the corresponding icon on the set you are using.

## Related publications

This section provides a list of additional documents referred to in this guide.

*Administration Guide* (NN40020-600)

*Device Configuration Guide* (NN40020-300)

*Installation Checklist and Quick Start Guide* (NN40020-308)

*Installation and Maintenance Guide* (NN40020-302)

*Main Office Configuration Guide for SRG50* (NN43001-307)

*Networking Configuration Guide* (NN40020-603)

*Telephony Device Installation Guide* (NN40020-309)

## How to get help

This section explains how to get help for Nortel products and services.

### Getting Help from the Nortel Web site

The best way to get technical support for Nortel products is from the Nortel Technical Support Web site:

<http://www.nortel.com/support>

This site provides quick access to software, documentation, bulletins, and tools to address issues with Nortel products. More specifically, the site enables you to:

- download software, documentation, and product bulletins
- search the Technical Support Web site and the Nortel Knowledge Base for answers to technical issues
- sign up for automatic notification of new software and documentation for Nortel equipment
- open and manage technical support cases

### Getting Help over the phone from a Nortel Solutions Center

If you don't find the information you require on the Nortel Technical Support Web site, and have a Nortel support contract, you can also get help over the phone from a Nortel Solutions Center.

In North America, call 1-800-4NORTEL (1-800-466-7835).

Outside North America, go to the following Web site to obtain the phone number for your region:

<http://www.nortel.com/callus>

### Getting Help from a specialist by using an Express Routing Code

To access some Nortel Technical Solutions Centers, you can use an Express Routing Code (ERC) to quickly route your call to a specialist in your Nortel product or service. To locate the ERC for your product or service, go to:

<http://www.nortel.com/erc>

### Getting Help through a Nortel distributor or reseller

If you purchased a service contract for your Nortel product from a distributor or authorized reseller, contact the technical support staff for that distributor or reseller.



# Chapter 2

## SRG50 overview

The SRG50 is a software application that leverages the BCM50 platform. It is optimized to provide feature transparency to the main office call server and to act as a survival remote gateway in a CS 1000 IP branch office environment.

SRG50 supports up to 80 survivable IP users with a single SRG application authorization code.

SRG50 Release 3.0 operates with CS 1000 running Release 4.0, 4.5, 5.0, and 5.5.

Configure SRG50 with Business Element Manager (Element Manager) or Network Configuration Manager (NCM). For detailed information about managing SRG50 with NCM, see your NCM documentation.

For a summary of differences between SRG versions, see the following table.

**Table 2** Summary of supported features on the different SRG versions (Sheet 1 of 2)

Features	SRG 1.5 on BCM 200/400	SRG50 1.0	SRG50 2.0	SRG50 3.0
Platform	Available on BCM 200/400 hardware. SRG 1.5 is not available on BCM1000 hardware.	Available on standard BCM50 hardware. SRG50 1.0 is not available on BCM50a or BCM50e hardware.	Available on standard BCM50 and BCM50b hardware only. SRG50 2.0 is not available on BCM50a, BCM50ba, BCM50e, and BCM50be hardware.	Available on standard BCM50 and BCM50b hardware only. SRG50 3.0 is not available on BCM50a, BCM50ba, BCM50e, and BCM50be hardware.
Call Servers	CS1000 Release 3.0, 4.0, and 4.5	CS1000 Release 3.0, 4.0, and 4.5	CS1000 Release 4.0, 4.5, and 5.0	CS1000 Release 4.0,4.5, and 5.0
Trunking	H323 is supported.	H323 is supported.	When CS1000 is supported by SRG, both H323 and SIP trunks are supported between them.	H323 and SIP trunking are supported.
IP clients	IP Phones 2001, 2002 (phase 1 and 2), 2004 (phase 1 and 2), 2007. IP KEM, IP Phone 2050 series. IP Phone 1110, 1120, 1140, IP Phone 2033 (polycom), WLAN 2210/2211/2212.	IP Phones 2001, 2002 (phase 1 and 2), 2004 (phase 1 and 2), 2007. IP KEM, IP Phone 2050 (v1). IP Phone 2033 (polycom), 2122	IP Phones 2001, 2002 (phase 1 and 2), 2004 (phase 1 and 2), 2007. IP KEM, IP Phone 2050 series. IP Phone 1120, 1140, IP Phone 2033 (polycom), WLAN 2210/2211/2212.	IP Phones 2001, 2002 (phase 1 and 2), 2004 (phase 1 and 2), 2007. IP KEM, IP Phone 2050 series. IP Phone 1110, 1120, 1140, IP Phone 2033 (polycom), WLAN 2210/2211/2212.
Number of IP clients supported	90	32	80	80
NetIQ	Not supported	Not supported	Supported	Supported
PVQM	Not supported	Not applicable	Supported	Supported

**Table 2** Summary of supported features on the different SRG versions (Sheet 2 of 2)

Features	SRG 1.5 on BCM 200/400	SRG50 1.0	SRG50 2.0	SRG50 3.0
ESA	Supported	Supported	Supported	Supported. On Site Notification (OSN) of E911 alarms supported for local users through a third party tool.
Vo911	Supported	Supported	Supported	Supported.
Enhanced firmware download	Supported	Supported	Supported	Supported
Analog station support	The number supported depends on the hardware. For BCM200: two analog station modules are supported. number of analog nodes= 2*8 =16. For BCM400: Four analog station modules are supported. number of analog nodes= 4*8 =32)	Two analog station modules are supported. number of analog nodes= 2*8 =16)	Two analog station modules are supported. number of analog nodes= 2*8 =16)	Two analog station modules are supported. number of analog nodes= 2*8 =16)

## SRG50 creation

An SRG50 is created by applying the appropriate SRG keycode to a functional BCM50 system. SRG50 is only supported on the BCM50 and BCM50b main units. Integrated router versions of the BCM50 (BCM50a, BCM50e, BCM50ba, and BCM50be) do not support the SRG50 application.

The *Installation Checklist and Quick Start Guide* is provided on the SRG50 Documentation CD that is shipped with your SRG50 system. Instructions in that guide are referenced in the following procedures. Also, the relevant BCM50 default IP addresses, user names, and passwords are excerpted from that guide and provided for your reference in the table [BCM50 default IP addresses](#) on page 16 and the table [BCM50 default user names and passwords](#) on page 17.

**Table 3** BCM50 default IP addresses

Port	IP address	Subnet mask
OAM port (see Note)	10.10.11.1	255.255.255.252
BCM50 LAN (no router)	192.168.1.2	255.255.255.0
<b>Note:</b> DHCP is enabled on the OAM port and assigns the following IP address: <b>10.10.11.2</b>		



**Table 4** BCM50 default user names and passwords

Tool	User ID/ User Name	Password
Element Manager	nnadmin	PlsChgMe!
Onbox main web page (http:// [IP address])	nnadmin	PlsChgMe!

## SRG50 keycode activation

To create an SRG50, use Element Manager to activate the SRG keycode on a BCM50 system (BCM50 or BCM50b main unit).

### To activate the SRG keycode

- 1 Locate the SRG authorization codes supplied with your product.
- 2 Open Element Manager. For information about installing and opening Element Manager, see the *Installation Checklist and Quick Start Guide* (NN40020-308).
- 3 With Element Manager, connect to the BCM system that you want to convert to an SRG. For information about connecting to a BCM system, see the *Installation Checklist and Quick Start Guide* (NN40020-308).

- 4 Navigate to the **Keycodes** panel (Configuration > System > Keycodes).

- 5 Click **Connect to Nortel Keycode Retrieval system** to obtain the keycode file for your system from the Nortel Keycode Retrieval System (KRS).

For more information about keycodes, see the *Keycode Installation Guide* (NN40010-301).

- 6 In the KRS, generate the keycode file for your system and save it on your management computer.

Make sure the SRG feature is included in your keycode as well as any other features you require for your system. To use the on-site notification for Emergency Services, you need to include a LAN CTE port in your keycode.

- 7 In Element Manager, return to the keycodes panel.

- 8 Click **Load File**.

- 9 Browse to the location on your management computer containing the keycode file for this system.

- 10 Select the keycode file, and then click **Open**.

The keycode file is applied.

- 11 Reboot your system to complete the creation of the SRG.

## To reboot the system

- 1 In Element Manager, navigate to the **Reset** panel (Administration > Utilities > Reset).
- 2 Click **Reboot BCM50 System**.

## To verify that the SRG has been successfully created

- 1 In Element Manager, navigate to the **Keycodes** panel (Configuration > System > Keycodes).  
In the Feature licenses table, verify that the status of the SRG keycode is ACTIVE.
- 2 Open the **Resources** folder (Configuration > Resources).  
Verify that there is a Survivable Remote Gateway panel.

## SRG50 and BCM50 features comparison

The table [Comparison of BCM50 and SRG50](#) on page 18 compares SRG50 and BCM50 features.

**Table 5** Comparison of BCM50 and SRG50 (Sheet 1 of 2)

Item	BCM50	SRG50
MBMs	See the <i>Installation and Maintenance Guide</i> (NN40020-302)	Recommended: ASM8+ (8 port Analog Station Module); DTM (Digital Trunk Module - 24 lines on either T1 or E1 or PRI); BRI (4 line BRI S/T Module); GATM4 (Global Analog Trunk MBM - 4 port); GATM8 (Global Analog Trunk MBM - 8 port) ADID4 (4 analog lines) ADID8 (8 analog lines) G4x16 (16 digital ports and 4 analog trunks); G8x16 (16 digital ports and 8 analog trunks) Supported for ATA connections: DSM16 (Digital Station Module - 16 ports); DSM32 (Digital Station Module - 32 ports); 4x16 Combo (16 digital ports, 4 analog trunks and 1 analog station); Does not support: DDIM (Digital Drop and Insert Mux) FEM (Fiber Expansion Module) DECT (Digital Enhanced Cordless Telecommunications)
Digital telephone sets	Yes	No
FCAPS	Yes	Yes, extended to include SRG-specific alarms and keycodes
Network Configuration Manager	Yes	Yes
Telset Administration	Yes	No
CS 1000 Geographic Redundancy	N/A	Yes
CS 1000 Network Bandwidth Management	N/A	Yes

**Table 5** Comparison of BCM50 and SRG50 (Sheet 2 of 2)

Item	BCM50	SRG50
CS 1000 Adaptive Network Bandwidth Management	N/A	Yes
CS 1000 Alternative Call Routing	N/A	Yes
CS 1000 Emergency Services Access	N/A	Yes
Firmware Download from main office call server	N/A	Yes (CS 1000 Release 4.0, 4.5, 5.0, and 5.5) (CS 1000 Release 4.0 requires patch MPLR22418)
SRG-specific features for interaction with a main office call server, including: Heartbeat detection of WAN recovery; IP telephone redirection to main office in Normal Mode; Local Mode IP telephone interface; H.323 Gateway to PSTN under control of main office call server	N/A	Yes

## Supported devices

The SRG50 Release 3.0 supports:

- IP Phones 1110, 1120E, and 1140E
- IP Phone 2001, 2002, 2004, 2007, and 2033
- IP Phone Key Expansion Module (KEM) – The IP Phone KEM is supported on an SRG with normal mode IP Phones. It does not function with local mode or test local mode IP Phones.
- IP Softphone 2050 v1/v2 and Mobile Voice Client (MVC) 2050
- WLAN Handsets 2210, 2211, and 2212
- WLAN Handsets 6120 and 6140
- WLAN Handsets 1210, 1220, and 1230
- Analog (500/2500 type) telephones
- Analog devices such as fax machines

**Note:** Throughout this document, the IP Phones in this list are referred to collectively as IP Phones.

The SRG50 is positioned primarily to support IP telephones and clients. However, analog devices can be supported using analog station modules (ASM), or by using an analog terminal adapter (ATA2) in conjunction with a digital station module (DSM). The SRG50 does not support digital or ISDN telephones.

## SRG50 terminology

The table [SRG50 terminology](#) on page 20 identifies SRG terms that may be unfamiliar to main office installers. They are provided to facilitate communications between SRG and main office personnel. In the table, the Element Manager path where the term appears is provided for reference and may not represent every appearance of the term.

**Table 6** SRG50 terminology (Sheet 1 of 2)

Term	Description
Port	<p>For telephony configuration (Configuration &gt; Telephony), a port is an internal number that identifies a physical termination point for a telephone set or a physical trunk.</p> <p>For the configuration of resources (Configuration &gt; Resources) and data services (Configuration &gt; Data Services), port is used in the context of the TCP/IP protocol suite.</p>
IP Terminal	<p>IP telephone</p> <p>Configuration &gt; Resources &gt; Telephony Resources &gt; IP &amp; App Sets</p>
Sets	<p>Can refer to actual telephones, or to the directory number (DN) assigned to the port to which a particular telephone is connected.</p> <p>Telephone Configuration &gt; Resources &gt; Telephony Resources &gt; IP &amp; App Sets</p> <p>Mapping DN to Telephone Configuration &gt; Telephony &gt; Sets</p> <p>DN Configuration &gt; Telephony &gt; Lines &gt; Target Lines &gt; Target Lines table &gt; Control Set and Prime Set columns</p>
Trunks	<p>Trunks refer to external facilities that are connected to the SRG and provide incoming and outgoing communication paths. Paths can be physical (examples: loop; PRI; T1) or virtual (VoIP trunks).</p> <p>Configuration &gt; Resources</p>
Loop trunk	An analog loop (FXO) that connects to the PSTN: a POTS line.
Lines	<p>A line is the generic term used for all communication paths, both internal and external.</p> <p>Configuration &gt; Telephony &gt; Lines</p>
Physical Lines	<p>Physical trunks.</p> <p>Configuration &gt; Telephony &gt; Lines &gt; Active Physical Lines (Lines 061 to 124)</p>
VoIP Lines	<p>VoIP trunks.</p> <p>Configuration &gt; Telephony &gt; Lines &gt; Active VoIP Lines (Lines 001 to 024)</p>

**Table 6** SRG50 terminology (Sheet 2 of 2)

Term	Description
Target Lines	<p>Target lines are internal, virtual paths between trunks and telephones for incoming calls (only). They provide flexibility in the way trunks and telephones can be associated: target lines can be used to direct an incoming call to one or more telephones, direct one or more trunks to one phone, or direct several trunks (in a line pool) to one or more phones. Target lines are assigned to DN(s). A target line triggers ringing voltage to the telephone(s) connected to the port(s) associated with the DN(s) that the target line is assigned to. (For example, if a unique target line is assigned to each DN, only one telephone rings when the DN is called. If several DNs are assigned to one target line, calling any of the DNs ring all of the associated phones.)</p> <p>Target lines are required for auto-answer trunks. Because VoIP lines are set internally to auto-answer, target lines are required for SRG operation.</p> <p>Element Manager provides two methods for assigning target lines to DNs.</p> <p>1) Configuration &gt; Telephony &gt; Sets &gt; All DNs &gt; All DNs table &gt; Details for DN subpanel &gt; Line Assignment tab</p> <p>or</p> <p>2) Configuration &gt; Telephony &gt; Lines &gt; Target Lines &gt; Target Lines table &gt; Details for Line subpanel &gt; Assigned DNs tab</p> <p>The first method provides a convenient way to assign the target line to the DN when the DN record is configured. The second method provides fields that allow incoming digit strings to be mapped to the DN.</p> <p>(Lines 125 to 268)</p> <p>For more information about target lines, see the <i>Networking Configuration Guide</i> (NN40020-603).</p>
Line pool	Several of the same type of trunk configured as one group: a trunk group.

## Coordination with the main office call server

Configuration of the SRG branch office requires datafill at both the SRG and the main office call server. Main office configuration drives SRG configuration, and Nortel recommends that the main office activities be concluded before undertaking SRG configuration.

For information, see [CS 1000 considerations](#) on page 33.

## SRG operating modes

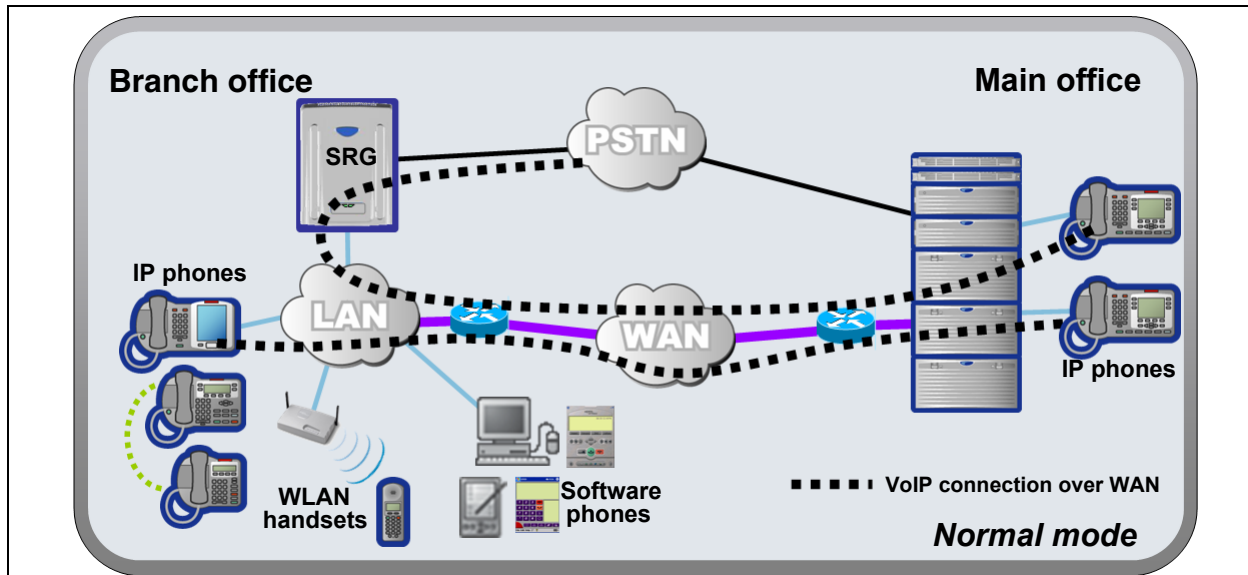
The SRG has two operating modes:

- Normal mode
- Local mode

### Normal mode

In normal mode (see the figure [Normal mode](#) on page 22), the SRG is connected to the main office call server over a WAN using VoIP trunks. From the perspective of the main office, the SRG is a branch office.

Figure 1 Normal mode



IP telephones connected at the SRG are registered with the main office call server and are under main office control. They operate as branch user sets and have access to all telephony services and features that the call server offers to IP telephones connected directly to the main office.


When a branch user set initiates a local PSTN call, the main office sets up the call using the VoIP trunks, which establishes a local media path. Emergency Services Access calls are similarly routed to the SRG PSTN. The telephone is redirected to local mode and the SRG initiates a local PSTN call to 911. For main office callers, the SRG acts as a VoIP-PSTN gateway during normal mode.

When call forwarding has been configured, incoming PSTN calls to the branch user set are forwarded over VoIP trunks (either H.323 or SIP) to the main office, which terminates the call at the branch user. Similarly, calls from analog telephones connected to the SRG to the branch user set are forwarded to the main office over VoIP trunks, which then terminates the call at the branch user. Calls from the branch user set to the analog telephones at the SRG are routed over the VoIP trunks to terminate at the analog telephone. In all these call scenarios, only signaling messages go through the VoIP trunk. The media path is set up directly between the branch user set and the voice gateway at the SRG. This means that these calls do not use any WAN bandwidth between the main office and the branch office after calls are established.

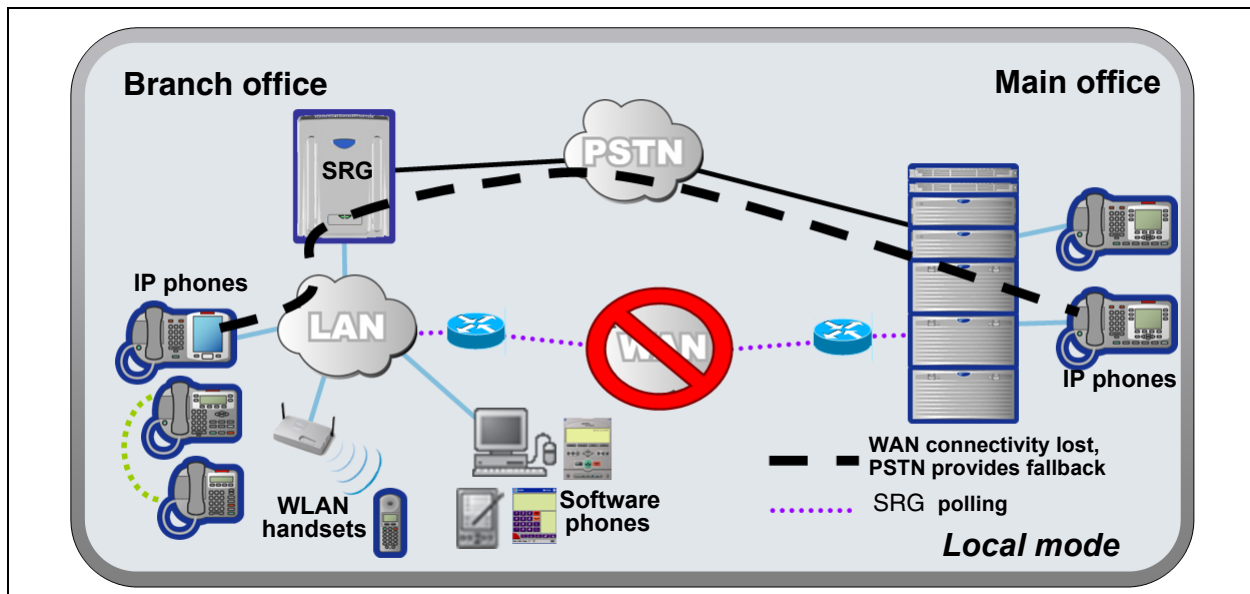
When a branch user IP telephone calls a main office IP telephone and vice versa, the call is a simple station-to-station call within the main office call server. Since the branch user IP telephone is physically remote from the call server, the media path goes through the WAN connection between the main office and the SRG, and thus uses WAN bandwidth, as demanded by the codec used in the call.

## Local mode

In the event of a WAN failure or the call server at the main office becomes unavailable, the IP Phones in normal mode revert to local mode automatically. In local mode, the IP users connected to the SRG are under the control of the SRG. When in local mode, main office call features are not available to users attached to the SRG. The SRG offers a set of basic features for the IP telephones, including access to the local PSTN, dialing emergency service numbers, and calling local extensions. For a complete list of local mode features, see [Features in local mode](#) on page 62. Local mode is illustrated in the figure [Local mode](#) on page 23.

 **Note:** The IP Phone KEM is supported on an SRG with normal mode IP Phones. It does not function with local mode or test local mode IP Phones.

**Figure 2** Local mode



The SRG handles all call processing. Calls between two IP telephones at the SRG are handled locally as a simple station-to-station call. When an IP telephone initiates a local PSTN call, the SRG routes the call to a trunk that is connected to the local PSTN. Incoming DID calls are also handled by the SRG and terminated on the appropriate IP telephone set.

In local mode, the IP telephones do not have access to the main office network over the VoIP trunks. If alternate routes are configured, then calls can be made to the main office or other branch offices using the available PSTN trunks.

Several situations, described below, can cause the IP phone to be in local mode.

### Initial registration

When the IP telephone is installed, it first registers with the SRG, and is in local mode. When the SRG configuration at the main office and the SRG is complete the IP telephone is redirected to the main office, where it registers as a branch user and changes from local mode to normal mode.

### **Automatic registration with the main office**

When configured as a branch office user set, an IP telephone at the SRG automatically attempts to register with the main office when:

- The phone is in local mode because of loss of connectivity with the main office, and the SRG is redirecting it back to the main office because connectivity has been reestablished (see [Loss of WAN or VoIP connectivity](#) on page 24).
- The phone is in local mode because Test Local Mode was invoked and the timer has expired or the Exit button is pressed.
- The phone is in local mode, the main office is a CS 1000, and this is the first time that the phone has been redirected to the main office.

The IP telephone can fail to register with the main office for several reasons. These are detailed in [Probable causes for redirection failure](#) on page 85.

### **Loss of WAN or VoIP connectivity**

The WAN or VoIP connectivity between the main office and the SRG can become unavailable if, for example, router failure occurs, the main office becomes unavailable, a WAN failure occurs, or the VoIP trunks reach capacity. When VoIP connectivity is lost, each IP telephone loses its connection with the main office terminal proxy server. The IP telephones reboot and reregister at the SRG, placing them in local mode. If enabled, call forwarding to the main office is automatically cancelled.

The IP telephones remain under the control of the SRG until VoIP connectivity is confirmed. When confirmation is received, the IP telephones are automatically redirected to the main office; redirection requires no user intervention. If the telephone is busy at the time that connectivity is reestablished, the SRG redirects the phone when it is free.

### **Test Local Mode**

Test Local Mode is a facility that allows the IP telephone to be redirected back to the SRG when it is in normal mode. This forces the IP telephone to go into local mode and allows the telephone user or system administrator to test local mode operation without taking down the VoIP trunks to the main office. Implementation of Test Local Mode depends on the main office call server. For more information, see [CS 1000 considerations](#) on page 33.

## **SRG installation and configuration summary**

The SRG50 Configuration Guide provides information specific to configuring a BCM50 as an SRG. Information pertaining to generic BCM50 practices and procedures is provided in the BCM50 documentation suite.

Generally, SRG50 activities leverage an installer's general knowledge of BCM50 activities. However, Nortel recommends that the BCM50/SRG50 site engineer and installer familiarize themselves with SRG-specific requirements before starting any installation activities.

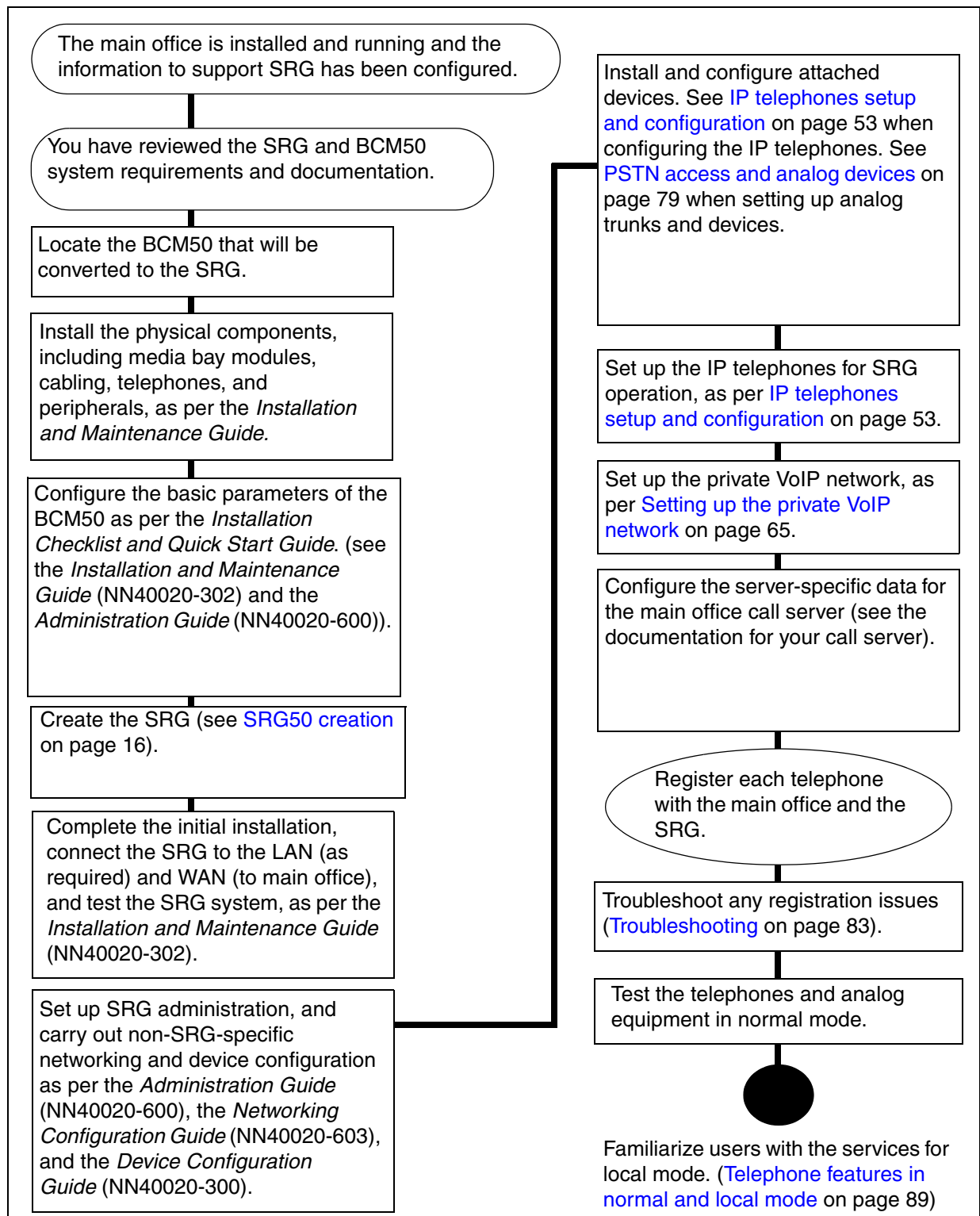


The figure [Process map for installing and configuring an SRG](#) on page 26 provides a process map for installing and configuring an SRG50. The procedures in this document assume that the following activities have been completed:

- The BCM50, including media bay modules, cabling, telephones, and peripherals, have been installed.
- BCM50 administration has been set up.
- The basic parameters of the BCM50 have been configured.
- CS1000 main office system has been installed and configured to support SRG.
- The SRG has been connected to the LAN (as required) and WAN (to the main office).
- System functionality has been tested to this point.
- Attached devices have been installed and configured (for information about configuring IP Phones, see [IP telephones setup and configuration](#) on page 53).
- Non-SRG-specific networking and device configuration has been completed (for information about configuring the network, see [Setting up the private VoIP network](#) on page 65).

## Process map for installing and configuring an SRG

**Figure 3** Process map for installing and configuring an SRG



# Chapter 3

## Task summary

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The task summary offers a high level, chronological review of the tasks required to configure the SRG50. The paths (**Xxxx > Yyyy > Zzzz**) direct you to the appropriate Business Element Manager (Element Manager) panels.

### Foundation configuration

Foundation configuration refers to configuration that is done as part of BCM50 foundation activities. The items identified here are significant for SRG operation and main office planning and installation.

- 1 Configure the SRG IP address, net mask, and gateway.  
**Configuration > System > IP Subsystem**  
 External Reference: *Installation and Maintenance Guide* (NN40020-302)
- 2 Confirm the number of IP sets and VoIP trunks.  
**Configuration > Resources > Application Resources**  
 The **Licence** column indicates the number of resources available.  
 External Reference: *Keycode Installation Guide* (NN40010-301)
- 3 Verify the global telephony settings.  
**Configuration > Telephony > Global Settings**  
 External Reference: *Device Configuration Guide* (NN40020-300)
- 4 Configure the Start DN (determined by the dialing plan).  
**Administration > Utilities > Reset > Cold Reset Telephony Services button > Cold Reset Telephony dialog box > Start DN field**  
 Internal Reference: [Basic parameters](#) on page 67  
 External Reference: *Installation and Maintenance Guide* (NN40020-302)
- 5 Verify the DN length.
  - i) For local calls between telephones on the SRG.  
**Configuration > Telephony > Dialing Plan > General > Dialing Plan - General panel > Global Settings subpanel > DN length (intercom) field**
  - ii) For incoming calls from the PSTN  
**Configuration > Telephony > Dialing Plan > Public Network > Dialing Plan - Public Network panel > Public Network Settings subpanel > Public Received number length field**

iii) For calls coming in from the private network

**Configuration > Telephony > Dialing Plan > Private Network > Dialing Plan - Private Network panel > Private Network Settings subpanel > Private Received number length field**

and

**Configuration > Telephony > Dialing Plan > Private Network > Dialing Plan - Private Network panel > Private Network Settings subpanel > Private DN length field**

(Private DN length is used for DPNSS applications only. See the *Networking Configuration Guide* (NN40020-603).)

Internal Reference: [Basic parameters](#) on page 67

External Reference: *Networking Configuration Guide* (NN40020-603)

- 6** Verify the line pool assignment of VoIP trunks.

In the default configuration, the VoIP trunks are assigned to line pool BlocA. Instructions in the SRG50 Configuration Guide assume that the default configuration has been maintained.

**Configuration > Telephony > Lines > Active VoIP Lines > Active VoIP Lines table > Line Type column > Line Type field**

External Reference: *Networking Configuration Guide* (NN40020-603)

- 7** The SRG supports four analog loop trunks on the main unit\*. Verify the line pool assignment of these trunks.

In the default configuration, these trunks are assigned to line pool A. Instructions in the SRG50 Configuration Guide assume that the default configuration has been maintained.

**Configuration > Telephony > Lines > Active Physical Lines > Active Physical Lines table > Line Type column > Line Type field**

External Reference: *Networking Configuration Guide* (NN40020-603)

\* Category 1 countries

## IP telephone configuration

- 1** Configure the registration password.

**Configuration > Resources > Telephony Resources > IP Sets row > Details for Module subpanel > IP Terminal Global Settings tab**

Internal Reference: [Registration password](#) on page 53

- 2** Configure the local mode indication (Advertisement/Logo).

**Configuration > Resources > Telephony Resources > IP Sets row > Details for Module subpanel > IP Terminal Global Settings tab**

Internal Reference: [Local mode indication](#) on page 55

- 3 Configure the IP telephone codec and jitter settings.  
**Configuration > Resources > Telephony Resources > IP Sets row > Details for Module subpanel > IP Terminal Global Settings tab**  
 Internal Reference: [IP telephone codec and jitter settings](#) on page 55
- 4 Configure the telephone (DN) records.  
**Configuration > Telephony > Sets > All DNs**  
 Internal References: [Telephone \(DN\) records configuration](#) on page 56
- 5 Configure the received numbers.  
**Configuration > Telephony > Lines > Target Lines**  
 Internal Reference: [Received numbers configuration](#) on page 59  
 External Reference: *Networking Configuration Guide* (NN40020-603)
- 6 Decide on the call forwarding option.  
 Internal Reference: [Call forwarding options](#) on page 60
- 7 Configure the IP telephones.  
 Internal Reference: [Configuration settings for redirected phones](#) on page 61

## Dialing plan configuration

- 1 Configure the private network type (CDP or UDP).  
**Configuration > Telephony > Dialing Plan > Private Network > Dialing Plan - Private Network panel > Private Network Settings subpanel > Private network type**  
 Internal Reference: [Private dialing plan](#) on page 68  
 External Reference: *Networking Configuration Guide* (NN40020-603)
- 2 Enable MCDN TAT.  
**Configuration > Telephony > Dialing Plan > Private Network > Dialing Plan - Private Network panel > MCDN subpanel**  
 Internal Reference: [Meridian Customer Defined Network \(MCDN\)](#) on page 69  
 External Reference: *Networking Configuration Guide* (NN40020-603)

## VoIP trunk configuration

Your SRG VoIP trunks are either H323 trunks, SIP trunks, or a combination of both types.

- 1 Configure routing table.  
**Configuration > Resources > Telephony Resources > Modules panel > IP Trunks row > Routing Table.**  
 External Reference: *Networking Configuration Guide* (NN40020-603)

- 2 Configure VoIP trunk QoS settings.

**Configuration > Resources > Telephony Resources > Modules panel > IP Trunks row > H323 Media Parameters tab or SIP Media Parameters tab**

Internal Reference: [QoS settings \(codec, jitter buffer, and related items\)](#) on page 70

External Reference: *Networking Configuration Guide* (NN40020-603)

- 3 Enable or disable fallback.

**Configuration > Resources > Telephony Resources > Modules panel > IP Trunks row > H323 Settings tab or SIP Settings tab**

Internal References:

[Fallback configuration](#) on page 72

[SRG PSTN access](#) on page 76

External Reference: *Networking Configuration Guide* (NN40020-603)

- 4 Configure H323 settings.

**Configuration > Resources > Telephony Resources > Modules panel > IP Trunks row > H323 Settings tab**

Internal Reference: [Gatekeeper routing](#) on page 73

External Reference: *Networking Configuration Guide* (NN40020-603)

- 5 Configure SIP settings.

**Configuration > Resources > Telephony Resources > Modules panel > IP Trunks row > SIP Settings tab**

External Reference: *Networking Configuration Guide* (NN40020-603)

- 6 Assign VoIP trunks to a line pool (if default configuration has not been maintained).

**Configuration > Telephony > Lines > Active VoIP Lines**

Internal Reference: [Line pools](#) on page 74

External Reference: *Networking Configuration Guide* (NN40020-603)

**Note:** In configuring SIP trunks, the SRG BCM shall register itself with the configured H323ID with NRS in CS1000. It is with this ID, the SRG will get MOTPS IP address from NCS.

- 7 Assign PSTN trunks to a line pool (if default configuration has not been maintained).

**Configuration > Telephony > Lines > Active Physical Lines**

Internal Reference: [Line pools](#) on page 74

External Reference: *Networking Configuration Guide* (NN40020-603)

- 8 Assign remote access packages to the VoIP trunks.

**Configuration > Telephony > Call Security > Remote Access Packages**

Internal Reference: [SRG PSTN access](#) on page 76

External Reference: *Networking Configuration Guide* (NN40020-603)

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## Call routing configuration

- 1 Decide on the fallback scheme.

Internal Reference: [Fallback configuration](#) on page 72

- 2 Configure the outgoing routes (VoIP and PSTN fallback).

**Configuration > Telephony > Dialing Plan > Routing**

Internal Reference: [Outgoing calls configuration](#) on page 74

External Reference: *Networking Configuration Guide* (NN40020-603)

- 3 Configure access to the SRG PSTN (for both local and tandem calls).

**Configuration > Telephony > Dialing Plan > Routing**

Internal Reference: [SRG PSTN access](#) on page 76

External Reference: *Networking Configuration Guide* (NN40020-603)

- 4 Configure for Network Bandwidth Management and Advanced Network Bandwidth Management.

Internal Reference: [Bandwidth management configuration: NBWM, ADBWM, and ACR](#) on page 40

External Reference: *Branch Office: Installation and Configuration* (553-3001-214) and *Data Networking for Voice over IP* (553-3001-160)

- 5 Configure for Alternative Call Routing.

Internal Reference: [Bandwidth management configuration: NBWM, ADBWM, and ACR](#) on page 40

External Reference: *What's New for Communication Server 1000 Release 4.5* (553-3001-015)

## Redirection and call forward configuration

- 1 Configure the main office settings.

**Configuration > Resources > Survivable Remote Gateway > S1000 Main Office Settings tab**

Internal Reference: [CS 1000 information for the SRG](#) on page 44

- 2 Configure the IP terminal settings.

**Configuration > Resources > Survivable Remote Gateway > S1000 IP Terminal Details tab**

Internal Reference: [IP telephones redirection](#) on page 47





# Chapter 4

## CS 1000 considerations

A Survivable Remote Gateway (SRG) extends CS 1000 features from the main office and provides a business continuity solution to one or more remote SRG locations (branch offices).

The *Main Office Configuration Guide for SRG50* (NN43001-307) provides information specific to the configuration of an SRG50 on the CS 1000. This guide is included on the SRG50 documentation CD for your reference. Access to other CS 1000 documentation may be required if personnel are not familiar with configuration of branch offices on the CS 1000.

The following activities are specific to SRG50 configuration when the main office call server is a CS 1000:

- [“CS 1000 and SRG terminology comparison” on page 33](#)
- [“Normal and local mode overview” on page 35](#)
- [“Virtual trunk capacity” on page 38](#)
- [“Vacant Number Routing \(VNR\)” on page 38](#)
- [“Bandwidth management” on page 38](#)
- [“Bandwidth management configuration: NBWM, ADBWM, and ACR” on page 40](#)
- [“Emergency Services Access \(ESA\) configuration” on page 42](#)
- [“CS 1000 information for the SRG” on page 44](#)
- [“IP telephones redirection” on page 47](#)
- [“Firmware upgrade” on page 51](#)

## CS 1000 and SRG terminology comparison

The table [Comparison of CS 1000 and SRG terms and contexts](#) on page 33 compares configuration-related terms and contexts of the CS 1000 and the SRG.

**Table 7** Comparison of CS 1000 and SRG terms and contexts (Sheet 1 of 2)

Term or Context	CS 1000	SRG
Dialing plan	on-net / off-net dialing	Private / Public network dialing
Type of number	CDP / UDP / GDP / TNDN	CDP / UDP / no equivalent
Numbers	TN (terminal number)	MOTN (main office terminal number)
	TN = MOTN. That is, the TN from the main office is entered on the SRG in the MOTN field (see <a href="#">“IP telephones redirection” on page 47</a> ).	
	BUID (branch user ID) The dialable number of an IP telephone at the SRG when it is called from a phone located at the main office or another branch office.	The CS 1000 BUID is entered on the SRG (see <a href="#">S1000 IP Terminal Details panel</a> on page 48) but there is no SRG equivalent for BUID.

**Table 7** Comparison of CS 1000 and SRG terms and contexts (Sheet 2 of 2)

Term or Context	CS 1000	SRG
	DN (directory number) The dialable number of a telephone at the main office when it is called from another phone at the main office.	DN (directory number) The dialable number of a telephone at the SRG when it is called from another phone at the SRG.
	<p>In the case of a CDP dialing plan, it is recommended that the BUID and the SRG DN be the same.</p> <p>In the case of a UDP dialing plan, the BUID has the form: &lt;VOIP Trunk Access Code&gt; + &lt;LOC&gt; + &lt;DN&gt;. In this case, it is recommended that the SRG DN be the same as &lt;DN&gt;.</p> <p>The dialable number of an IP telephone at the SRG, when dialed from another phone at the SRG, remains the same in both normal and local mode if the preceding recommendations are implemented.</p>	
	AC1	VOIP Trunk Access Code (see <a href="#">CS 1000 information for the SRG</a> on page 44) Destination code for VoIP trunks (see <a href="#">Outgoing calls configuration</a> on page 74)
	AC1 = VOIP Trunk Access Code = Destination code for VoIP trunks	
Routing	distant steering codes (DSC), trunk steering codes (TSC), local steering codes (LSC)	call routing, destination codes, line pool access codes
	digit manipulation table	dialout digits (routing)
Numbering Plan ID	ISDN/Telephony (E.164), Private, Telephony (E.163), Telex (F.69), Data (X.121), National Standard	Private
Access codes (SRG: Destination codes)	7 = system trunk access 8 = Basic Alternate Route Selection (BARS)/Network Alternate Route Selection (NARS) 9 = public exchange access	7 = not assigned 8 = not assigned 9 = line pool A access code
Network Class of Service	Facility Restriction Level (FRL)	scheduled call routing
Network Bandwidth Management	Zone ID	Zone ID
		Virtual Private Network ID (VPNI)
	<p>CS 1000 Zone ID = SRG Zone ID ZDP = VPNI.</p> <p>That is, the CS 1000 Zone ID is entered on the SRG in the Zone ID field, and the ZDP is entered on the SRG in the VPNI field (see <a href="#">Bandwidth management</a> on page 38).</p>	
Trunks	public exchange	PSTN
IP telephone password	installer password	global password
	The two passwords can be made the same. See <a href="#">Registration password</a> on page 53.	

## Normal and local mode overview

Normal mode and local mode overview provides a description of the following sections:

- “Normal mode” on page 35
- “Local mode” on page 35
- “Survivability” on page 36
- “Recovery to normal mode” on page 37
- “Testing the telephone in local mode” on page 37

### Normal mode

IP Phones that are physically located at the SRG but are registered with the main office are operating in normal mode. In normal mode, the main office provides centralized call processing to all applications transparently to all IP Phones at the Branch Office.

Users of the SRG IP Phones receive the features, applications, key layout, and tones of the main office Call Server. This provides feature and application transparency between the branch office and the main office.

### Local mode

Users at the branch office may be in local mode, or survivable mode for two different reasons:

- 1 IP Phone may have just booted up.
- 2 IP Phone cannot communicate to the main office because of a WAN failure or a failure of the main office components.



**Note:** When a telephone or trunk in the main office calls an SRG IP Phone that has switched to local mode due to WAN failure, the call is treated according to the main office call redirection configuration (such as forwarding to voice mail or continuous ringback).

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In the event that the IP Phones at the branch office lose the connection to the main office CS1000 call server for any reason (WAN failure, main office call server failure, main office Signaling Server failure), the SRG50 reverts to local mode automatically. Essentially, when VoIP connectivity is lost, each IP Phone loses its Reliable UDP (RUDP) connection with the main office terminal proxy server (TPS). The IP Phones at the branch office reboot and re-register to the SRG50, placing them in local mode.

Once this has occurred, the IP Phones displays an indication on the display area that the set is in local mode of operation. This display is configurable by installers to meet local language and usage norms.

In local mode, the IP users connected at the branch office are under the control of the SRG50 call services. As such, the normal main office call server features are not available. The SRG50 offers a basic feature set when in local mode which allows IP Phones to continue to make and receive calls internally within the branch office and over the provisioned local PSTN interfaces. Basic

services, such as transfer, last number redial, and single key access through the PSTN to a centralized voice messaging system are supported. Local PSTN access and local Emergency Services access is also supported. No local applications or Business Communication Manager features are supported in local mode operation.

Analog devices continue to be under the control of the SRG50 system. It is the intent of local mode to provide continued access to the PSTN for critical calls and emergency services.

In local mode, since the SRG50 handles all call processing, calls between two IP phones at the SRG50 are handled locally as a simple station-to-station call. When an IP Phone initiates a local PSTN call, the SRG50 routes the call to a trunk that is connected to the local PSTN. Incoming DID calls are also handled by the SRG50 and terminated on the appropriate IP Phone.

In the event of a WAN failure, in local mode, the IP Phones do not have access to the main office network over the VoIP trunks. If the appropriate alternate routes are configured, calls will be routed to the main office or other branch offices using the available PSTN trunks.

While in local mode, the SRG50 system continues to monitor for a main office CS1000 heartbeat signal, and once detected, automatically redirects phones on an individual basis back to normal mode of operation. If a call is active, the SRG waits until the call is completed before redirecting the phones; calls in progress are not interrupted. This redirection occurs almost immediately. This reinstates the CS1000 normal user interface and feature set for the IP Phone user, on a user by user basis.

The SRG50 system implements the same interface used by the MG1000B system to interact with the main office CS1000 system. This allows the main office to identify attached clients and the local PSTN as branch office entities, enabling proper operation of dial plans and E911 access.

In local mode, devices that are physically located at the branch office, that are controlled by the local system and receive a basic telephony feature set, provide business continuity for the branch office during the WAN or system failure. The SRG supports a main office heartbeat which automatically reregisters users once WAN or system failure has recovered.

## Survivability

SRG is specifically designed to provide automatic survivability against WAN failure, main office Call Server failure, main office Signaling Server failure, and NRS (gatekeeper) failure.

SRG supports the Geographic Redundancy feature.

In the event of a WAN failure, the SRG IP Phones lose communication with the main office. This causes the SRG IP Phones to reset and register with the SRG. The IP Phones then operate in local mode, providing basic telephony services delivered by the local SRG system.

If the main office Call Server fails and call processing services are provided by an Alternate Call Server, the SRG IP Phones reset and reregister with the Alternate Call Server and receive call processing services from it. If no Alternate Call Server is available, the SRG IP Phones go to local mode while the SRG attempts to find an Alternate Call Server by way of the network connection server (NCS).

If the main office Signaling Server fails and an Alternate Signaling Server is available, the SRG IP Phones reset and reregister with the SRG. The SRG will then query the NCS for the Alternate Signaling Server's IP address. The SRG will redirect the IP Phone to the Alternate Signaling Server and continue to receive call processing services from the main office Call Server. If no Alternate Signaling Server is available, the SRG IP Phones reset and register with the SRG in local mode.

When an IP Phone at the SRG first boots up, the IP Phone attempts to communicate with the SRG. After communication with the SRG is established, the SRG redirects the IP Phone to the main office. When the SRG IP Phone attempts to register with the main office, the SRG first queries the Primary NRS for the main office Virtual Trunk node IP address to redirect the IP Phone. If the Primary NRS is down or unreachable, the SRG queries the Alternate NRS, if one is specified. If it receives a positive response, the SRG IP Phone is redirected to the specified main office. Otherwise, if neither a Primary or an Alternate NRS is available, the SRG IP Phone remains in local mode, and receives call processing services from the SRG until communication can be reestablished.

SRG IP Phones in normal mode remain registered with the main office if the Primary NRS fails and no Alternate NRS is available. They can call any main office telephone or IP Phones in normal mode in other branch offices.

SIP/ H323 trunks are used for Voice traffic alone. Proprietary TPSAR protocol is used for WAN link health check and for Main Office TPS address discovery through NRS.

However, they cannot call any SRG analog (500/2500-type) telephones or any external numbers through the SRG trunks because the Virtual Trunks are not available. (SRG analog [500/2500-type] telephones are accessible if alternate routing is available through the PSTN.)

For more information about virtual trunks for SRG50, see the *Main Office Configuration for Survivable Remote Gateway 50* (NN43001-307).

## Recovery to normal mode

Once communication is re-established with the main office call server, all IP Phones at the branch office that are in local mode automatically redirect and reregister to the main office and return to normal mode operation. IP Phones that were busy at the time communication was reestablished complete the call in local mode, and then reregister with the main office after the call is complete.

## Testing the telephone in local mode

From normal mode, the branch user has the option of going to local mode manually using the Test local mode feature. The test can be performed by the user at any time and does not require a password. This test is invoked from any IP Phone at the branch office.

Nortel recommends testing local mode operation after changing the provisioning for a telephone on the SRG.

To ensure that users do not forget to resume normal mode operation, the SRG redirects the telephone to the main office to return the telephone to normal mode. This occurs if the telephone remains registered to the SRG in test local mode for ten minutes (default setting). Alternatively, the user can press the Quit key from the set to return to normal mode.

## Virtual trunk capacity

The SRG supports a number of simultaneous calls depends on the specific codec type used.

SRG50 supports 24 virtual IP trunks that are either H.323 only, SIP only, or a combination of the H.323 and SIP.

In normal mode, the codec selection used is controlled by specific programming of the CS1000. In this case: SRG50 supports up to a maximum of 15 Virtual trunks unless both the intrazone and interzone codecs are configured as Best Quality (G.711) in which case, the maximum number of virtual trunks would be 24.

In local mode, if the WAN has failed, there are no longer any virtual trunks available between the SRG50 and CS1000. However, the SRG50 will continue to convert calls from IP terminals for communication via the PSTN. In this case, if G.711 is used (recommended), the number of simultaneous calls from IP terminals to the PSTN supportable is a maximum of 24.

For more information about virtual trunks for SRG50, see the *Main Office Configuration for Survivable Remote Gateway 50* (NN43001-307).

## Vacant Number Routing (VNR)

The SRG does not support Vacant Number Routing (VNR). Instead, the SRG uses Call Forward All Calls to emulate VNR for the IP telephones that are in normal mode. Call Forward All Calls is automatically cancelled when the phones revert to local mode.

A single destination code and route (or a group of destination codes and routes) can be configured on the SRG to route all calls not terminated locally by the SRG. These calls are routed over the VoIP trunks. If the VoIP trunks become unavailable, the calls are routed to the proper location using PSTN fallback. This feature is similar to the VNR feature on the Media Gateway 1000B (MG1000B).

Seamless dialing requires that the start digit of the DNs are unique for each system (coordinated dialing plan). If the start digit is the same on both systems, the local users on the SRG must dial a separate destination code before the main office DN.

For details about dialing plan and routing configuration, see [Setting up the private VoIP network](#) on page 65.

## Bandwidth management

Three levels of bandwidth management are supported by the CS 1000:

- Network Bandwidth Management (NBWM)
- Adaptive Network Bandwidth Management (ADBWM)
- Alternative Call Routing (ACR)

## Network Bandwidth Management (NBWM)

The SRG interoperates with the Network Bandwidth Management (NBWM) feature in a manner similar to Media Gateway (MG) 1000B, though only G.711 and G.729 codecs are supported. At the SRG, a Virtual Private Network ID (VPNI) and Zone ID are entered with values defined by the main office configuration (see [Bandwidth management configuration: NBWM, ADBWM, and ACR](#) on page 40). The VPNI and Zone ID allow the CS 1000 to recognize that H.323 and SIP calls to and from the SRG are from a specific Bandwidth Management zone.

NBWM allows bandwidth zones to be configured on a network basis so that codec selection and bandwidth allocation software can identify whether IP telephones or gateways are physically collocated (in the same bandwidth zone) even though they are controlled by different call servers. NBWM is used to define the codec selection policy and track bandwidth used for calls that traverse the WAN (interzone calls) and the LAN (intrazone calls). The bulk of configuration for NBWM is done at the main office.

## Adaptive Network Bandwidth Management (ADBWM)

As with NBWM, only the VPNI and Zone ID are required at the SRG in order to implement the Adaptive Network Bandwidth Management (ADBWM) feature on the SRG (see [Bandwidth management configuration: NBWM, ADBWM, and ACR](#) on page 40).

ADBWM uses real-time interaction to enhance the performance of Voice over Internet Protocol (VoIP) networks. ADBWM adjusts bandwidth limits and takes corrective action in response to Quality of Service (QoS) feedback. This adjustment occurs dynamically, while calls are in progress. A call server with ADBWM uses VPNI and Zone IDs to keep track of the bandwidth being used between its own zone and zones belonging to other call servers. If the interzone QoS degrades below an acceptable level, the available bandwidth is reduced automatically between the two zones. When the QoS between the two zones improves, the bandwidth limit is allowed to return to normal.

## Alternative Call Routing (ACR)

Configuration for Alternative Call Routing (ACR) at the SRG includes datafilling the Virtual Private Network ID (VPNI) and Zone ID required by NBWM and ADBWM. However, additional configuration is required and depends on the type of trunking provisioned at the main office: Attendant service or DID trunks (see [Bandwidth management configuration: NBWM, ADBWM, and ACR](#) on page 40).

ACR for NBWM allows a station-to-station call (that is, a call that does not use a trunk) to overflow to traditional routes. Overflow can occur if there is insufficient interzone bandwidth available to carry the call, or if the QoS has degraded to unacceptable levels. The feature applies to station-to-station calls between a branch office and main office as well as from one branch office to another branch office, provided both stations are registered to the same main office.

Network administrators who do not want calls to be blocked, yet have a limited amount of bandwidth available, can use ACR to overflow calls to conventional trunks (PSTN or Tie/MCDN). ACR allows calls to be routed by overflowing them, trading off the capital cost of WAN bandwidth against the incremental cost of overflowed calls.

## Bandwidth management configuration: NBWM, ADBWM, and ACR

### To configure SRG for NBWM and ADBWM

- 1 Obtain the Virtual Private Network ID and the Zone ID numbers configured at the main office.
- 2 Use Element Manager to enter these numbers in the appropriate fields at Configuration > Telephony > Dialing Plan > Private Network > Dialing Plan - Private Network panel > VoIP subpanel (see the figure [Dialing Plan - Private Network panel, VoIP subpanel](#) on page 40).

**Figure 4** Dialing Plan - Private Network panel, VoIP subpanel

The screenshot displays the 'Dialing Plan - Private Network' configuration panel, specifically the VoIP subpanel. On the left is a 'Task Navigation Panel' with a tree view showing the configuration hierarchy. The main area is titled 'Dialing Plan - Private Network' and contains several sections:

- Private Network Settings:** Includes fields for 'Private Received number length' (set to 3), 'Private Auto DN', 'Private DISA DN', 'Private access code', 'Private network type' (set to None), 'Location code', and 'Private DN length' (set to 3).
- MCDN:** Includes fields for 'Local access code', 'National access code', 'Special access code', and checkboxes for 'Network ICCL', 'TRO' (checked), and 'TAT'.
- VoIP:** Includes input fields for 'Virtual Private Network ID' (set to 0) and 'Zone ID' (set to 0).
- ETSI:** Includes checkboxes for 'Network Diversion' and 'MCID'.

A red arrow points from the 'Private network type' dropdown menu to the 'Virtual Private Network ID' and 'Zone ID' input fields in the VoIP section.

### To configure Alternative Call Routing with attendant service

- 1 Complete the procedure, [To configure SRG for NBWM and ADBWM](#) on page 40.  
For more information, see [Outgoing calls configuration](#) on page 74, and see also the *Networking Configuration Guide* (NN40020-603).
- 2 Obtain the ALTPrefix for the SRG (configured at the main office).



- 3 Define a route to the main office Attendant over the PSTN.
  - a Access the **Dialing Plan - Routing** panel (**Configuration > Telephony > Dialing Plan > Routing**) and select the **Routes** tab.
  - b Add a new route (for example, 997).
  - c Ensure that the **DN Type** is **Public (Unknown)**.
  - d In the **External Number** field, enter the **PSTN number** of the main office Attendant telephone.
  - e Assign the PSTN line pool to the route (select the line pool from the **Use Pool** list; default is **A**).
- 4 Add a destination code.
  - a Access the **Dialing Plan - Routing** panel (**Configuration > Telephony > Dialing Plan > Routing**) and select the **Destination Codes** tab.
  - b Add a new destination code.  
Use the ALTPrefix as the destination code.
  - c In the **ALTPrefix Destination Code** row, select the **Normal Route** field.
  - d Enter the route for the Attendant telephone (997).
  - e In the adjacent **Absorbed Length** field, select **All** from the list.

When the SRG receives the ALTPrefix+DN digits from the main office, it looks up the destination code table, finds a match for the ALTPrefix, dumps all the digits (ALTPrefix+DN), and dials the main office Attendant.

## To configure Alternative Call Routing with DID trunks



**Note:** In order to support DID trunks, the MODN dialed by an SRG caller must match the DID digits.

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**Note:** In the following procedure, it is assumed that the MODN matches the XXXX portion of the DID's NPA-NXX-XXXX; and that the ALTPrefix is 3 digits.

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- 1 Complete the procedure, [To configure SRG for NBWM and ADBWM](#) on page 40.  
For more information, see [Outgoing calls configuration](#) on page 74 and the *Networking Configuration Guide* (NN40020-603).
- 2 Obtain the ALTPrefix for the SRG (configured at the main office).
- 3 Define a route for the NPA-NXXX portion of the main office DID numbers.
  - a Access the **Dialing Plan - Routing** panel (**Configuration > Telephony > Dialing Plan > Routing**) and select the **Routes** tab.
  - b Add a new route (for example, 996).

- c Ensure that the **DN Type** is **Public (Unknown)**.
  - d In the **External Number** field, enter the NPA-NXXX of the DID trunks that serve the main office.
  - e Assign the PSTN line pool to the route (select the line pool from the **Use Pool** list; default is **A**).
- 4** Add a destination code.
- a Access the **Dialing Plan - Routing** panel (**Configuration > Telephony > Dialing Plan > Routing**) and select the **Destination Codes** tab.
  - b Add a new destination code.  
Use the **ALTPrefix** as the destination code.
  - c On the **ALTPrefix Destination Code** row, select the **Normal Route** field.
  - d Enter the route for the route added above (996).
  - e In the adjacent **Absorbed Length** field, select **3** from the list.

When the SRG receives the ALTPrefix+DN digits from the main office, it looks up the destination code table, finds a match for the ALTPrefix, dumps the 3- digit ALTPrefix, appends the DN to the **External Number** and dials the **External Number+DN**.

## Emergency Services Access (ESA) configuration

The *Main Office Configuration for Survivable Remote Gateway 50* (NN43001-307) guide covers the procedures for configuring Emergency Services Access on both the SRG and the CS 1000. The information here expands on the SRG procedure in that guide.

There are two methods of ESA support:

- When a user dials 911, the telephone is redirected to the Branch Office and connects to the SRG in local mode. The Main Office indicates the reason for redirection (as e911) in the DRAM. The SRG dials 911 across the PSTN. The telephone remains in local mode after the 911 call and is redirected to the Main Office after 10 minutes.
- On-site notification for 911 — An external tool records an alarm when 911 calls are made at the branch office. This is applicable to both telephones that are in local mode due to the activation of a 911 call as well as locally connected analog sets. The external tool connects to the SRG through LAN CTE.



**Note:** This procedure applies only to redirected IP telephones when the SRG is in normal mode.

For IP telephones in local mode, and for other telephones at the SRG, see the *Networking Configuration Guide* (NN40020-603) for configuring emergency services.

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## To configure Emergency Services Access

- 1 Verify that a remote access package has been assigned to the VoIP trunks (see [Remote Access Package for VoIP trunks](#) on page 76).
- 2 Obtain the ESA Special Number (SPN).
- 3 In Element Manager, access the **Dialing Plan - Routing** panel (**Configuration > Telephony > Dialing Plan > Routing**) and click the **Destination Codes** tab.
- 4 Add a destination code corresponding to the ESA SPN for the SRG branch office.
- 5 In the **Destination Codes** table, click the **Absorbed Length** field of the ESA SPN destination code. The numbers indicate the number of digits the SRG absorbs, from left to right.
- 6 Select the number of digits to absorb so that just the Emergency Services DN (ESDN) remains.
- 7 In the **Destination Codes** table, click the **Normal Route** field of the ESA SPN destination code. Enter a public route to the PSTN trunks.



**Note:** The **Normal Route** field defaults to 000. Route 000 (click the **Routes** tab) is preconfigured to Use Pool A and cannot be changed. Pool A is preconfigured for PSTN trunks in the default state. Hence, if the default state of Pool A has not been changed, leave the Normal Route field as 000.

To check the state of Pool A, navigate to **Configuration > Lines > Active Physical Lines**. Pool A must be assigned to at least one Trunk Type that provides access to the PSTN.

Do not configure **Alternate Routes**.

- 8 Navigate to the Dialing Plan - Public Network panel (**Configuration > Dialing Plan > Public Network**).
- 9 In the **Public Network DN Lengths** subpanel, verify that there is a **DN Prefix** of 911 with a **DN Length** of 3.  
If not, add the 911 **DN Prefix**. If required, double click the **DN Length** field and then change the value to 3.

## To configure on-site notification

To use the on-site notification for Emergency Services, you need to include a LAN CTE port in your keycode.

## CS 1000 information for the SRG

In order to redirect IP telephones and forward calls to the main office (Call Forward All Calls feature), the SRG requires information about the main office network environment. This information is recorded through Element Manager on the S1000 Main Office Settings panel.

### S1000 Main Office Settings panel

The table [S1000 Main Office Settings](#) on page 44 lists and describes each field of the S1000 Main Office Settings panel. Record the actual value in the Values column to facilitate configuration and provide a record of the datafill.

**Table 8** S1000 Main Office Settings (Sheet 1 of 3)

Field	Values	Description
<b>Primary Network Connect Server Address</b>	<ip address>	IP address of the primary NCS.
<b>Alternate Network Connector Server Address</b>	<ip address>	IP address of the alternate NCS, if deployed. If not, enter the same address as for Primary Network Connect Server Address.
<b>Network Connect Server Port</b>	16500 (default) Range: 0 to 65535	Port on the SRG used to connect to the NCS.
<b>Heartbeat Protocol Port</b>	16501 (default) Range: 0 to 65535	Port on the SRG that the SRG uses to monitor the status of the connection with the main office terminal proxy server (that is, to confirm connectivity with the main office)
<b>VOIP Trunk Access Code</b>		Access code for the main office VoIP trunk. Required for UDP dialing plan only. Ignored for CDP dialing plan, field can be left blank. VOIP Trunk Access Code = Destination code for VoIP trunks* = AC1** * Destination code for VoIP trunks is entered during configuration for advanced routing. see <a href="#">“Outgoing calls configuration” on page 74</a> . ** For a UDP dialing plan, AC1 is the access code in the digit string <AC1> <LOC> <DN>
<b>Test Local Mode Timeout</b>	10 minutes (default) Range: 2 to 10 minutes	Period that an IP telephone remains in local mode after being set in local mode manually. Telephone returns to normal mode automatically at the end of the time-out.  Local mode can be invoked by the Test Local Mode button on the telephone or by command from the main office.

**Table 8** S1000 Main Office Settings (Sheet 2 of 3)

Field	Values	Description
<b>H323 ID</b>	SRG* (default) *This setting must be changed.  see the <i>Networking Configuration Guide</i> (NN40020-603) for naming conventions.	Gatekeeper setting that identifies the SRG.  This value must match the value in the <b>Alias names</b> field of the Local IP gateway: <b>Configuration &gt; Resources &gt; Telephony Resources &gt; Modules</b> panel > <b>Module type</b> column: select <b>IP Trunks &gt; Details for Module: Internal details</b> panel > <b>Local Gateway</b> tab > <b>Gatekeeper Support</b> subpanel > <b>Alias names</b> field (see <a href="#">VoIP trunking configuration</a> on page 72).  The SRG acts as a SIP endpoint and obtains the Main Office TPS IP address from the NCS using the H323 ID.  Configure this field as the same SIP endpoint name as you configure for the NRS endpoint name. This is a generic field used for both SIP and H.323 virtual trunk configurations.
<b>Numbering Plan ID</b>	Unknown ISDN/Telephony (E.164) Private Telephony (E.163) Telex (F.69) Data (X.121) National Standard  Default: Private	The type of numbering plan at the main office.
<b>Type of Number</b>	Unknown International Number National Number Special Number Subscriber Number ESN LOC (UDP) ESN CDP ESN Special Number  Default: ESN CDP (for CDP dialing plans) (BUID = DN) UDP dialing plans: select ESN LOC (UDP) (BUID = LOC+DN)	The main office dialing plan.  Ensure that the SRG private dialing plan is configured to match the selected value.

**Table 8** S1000 Main Office Settings (Sheet 3 of 3)

Field	Values	Description
<b>Node ID</b>	9999 (default) Range: 0 to 9999	Automatically written to the IP telephone firmware when the IP telephone registers with the main office. Used to identify the node on the main office associated with the IP telephone DN.
<b>MO Access Code Length</b>	For CDP dialing plans: set to 0 For UDP dialing plans: set to length of line pool access code or destination code in front of LOC. Range: 0 to 34	The number of digits to add to the BUID (DN) so the main office system can determine if the incoming call is valid.

## To datafill the S1000 Main Office Settings panel

- 1 In Element Manager, navigate to **Configuration > Resources > Survivable Remote Gateway** (see the figure [S1000 Main Office Settings panel](#) on page 46).
- 2 Select the **S1000 Main Office Settings** tab.
- 3 Enter the information in the appropriate fields.

**Figure 5** S1000 Main Office Settings panel

The screenshot shows the 'Survivable Remote Gateway' configuration page. The left sidebar is titled 'Task Navigation Panel' and has two tabs: 'Configuration' and 'Administration'. Under 'Configuration', there is a tree view with folders for 'System', 'Administrator Access', 'Resources', 'Telephony', and 'Data Services'. The 'Resources' folder is expanded, showing sub-items: 'Application Resources', 'Media Gateways', 'Port Ranges', 'Telephony Resources', 'Survivable Remote Gateway' (highlighted), and 'Network Interfaces'.

The main configuration area is titled 'Survivable Remote Gateway' and has two tabs: 'S1000 Main Office Settings' (active) and 'S1000 IP Terminal Details'. The settings are as follows:

- Primary Network Connect Server Address: 10.10.10.10
- Alternate Network Connect Server Address: 10.10.10.10
- Network Connect Server Port: 16500
- Heartbeat Protocol Port: 16501
- VOIP Trunk Access Code: (empty)
- Test Local Mode Timeout: 10
- H323 ID: SRG
- Numbering Plan ID: Private (dropdown)
- Type of Number: ESN CDP (dropdown)
- Node ID: 9999
- MO Access Code Length: 0

## IP telephones redirection

Once an IP telephone at the SRG is configured (see [IP telephones setup and configuration](#) on page 53), it automatically registers with the SRG (S1). To configure an IP telephone for redirection to the main office call server, SRG-specific datafill is required. The SRG-specific configuration includes:

- [IP telephones numbers and models](#) on page 47
- [S1000 IP Terminal Details panel](#) on page 48

To configure SRG using SIP trunks, CS1K must accept the end point as an H323 and SIP entity and accept it as h323 and SIP endpoint. You do not need to enter BCM SRG50 configuration under Telephony Resources > IP Trunks > H323 Setting. The entire configuration for SIP trunking must be entered under SIP settings, SIP Proxy, SIP media parameters, SIP URI Maps and SIP Authentication.

## IP telephones numbers and models

SRG DNs are assigned to IP telephones using Element Manager. Redirection to the main office requires configuration at the SRG to associate the SRG DN with the CS 1000 terminal number (TN) and the corresponding branch user ID (BUID).

When the CS1000 TN is configured, the IP telephone model forms part of the record. At the SRG, the actual IP telephone configured to an SRG DN must be the same model that is configured in the TN record that is associated with the SRG DN.

The table [IP telephone numbers and models](#) on page 47 provides a record of the CS 1000 TN and BUID, the SRG MOTN and DN, and the model of phone, to facilitate installation and provide a permanent record of the configuration.

**Table 9** IP telephone numbers and models (Sheet 1 of 2)

TN (CS 1000) MOTN (SRG) (same number)	BUID (same number at CS 1000 and SRG)	SRG DN	IP telephone model

**Table 9** IP telephone numbers and models (Sheet 2 of 2)

TN (CS 1000) MOTN (SRG) (same number)	BUID (same number at CS 1000 and SRG)	SRG DN	IP telephone model

## S1000 IP Terminal Details panel

Element Manager provides SRG-specific panels for recording the CS 1000 TN and BUID that are associated with a particular SRG DN. The table [SRG S1000 IP Terminal Details fields](#) on page 48 lists and describes the fields on the **S1000 IP Terminal Details** panel.

**Table 10** SRG S1000 IP Terminal Details fields (Sheet 1 of 2)

Field	Values	Description
<b>DN</b>	Read-only	The SRG DN assigned to the telephone. The SRG DN must be configured before proceeding with the procedures that follow in this section. See <a href="#">Telephone (DN) records configuration</a> on page 56.
<b>Hardware ID</b>	Read-only	Hardware ID. Unique for each IP telephone.
<b>Status</b>	Read-only	Current status of the telephone. See <a href="#">IP terminal details</a> on page 84 (expand the field to read the entire status message).



**Table 10** SRG S1000 IP Terminal Details fields (Sheet 2 of 2)

Field	Values	Description
<b>Firmware Version</b>	Read-only	Updated by the main office when a terminal is sent back to the SRG for firmware upgrade purposes. The field specifies the firmware version required by the main office.
<b>MOTN</b>	XXX	Required for telephone redirection. The field is the main office TN associated with the IP telephone.
<b>BUID</b>	CDP network: <DN>  UDP network: <VoIP access code> + <LOC> + <DN>	Required for telephone redirection. The field represents the dialable number of an IP telephone at the SRG when it is called from a phone located at the main office or another branch office. The BUID at the SRG must be the same as the BUID at the main office.
<b>MO TPS</b>	Read-only	This field echoes the address of the main office terminal proxy server when the IP telephone is redirected.



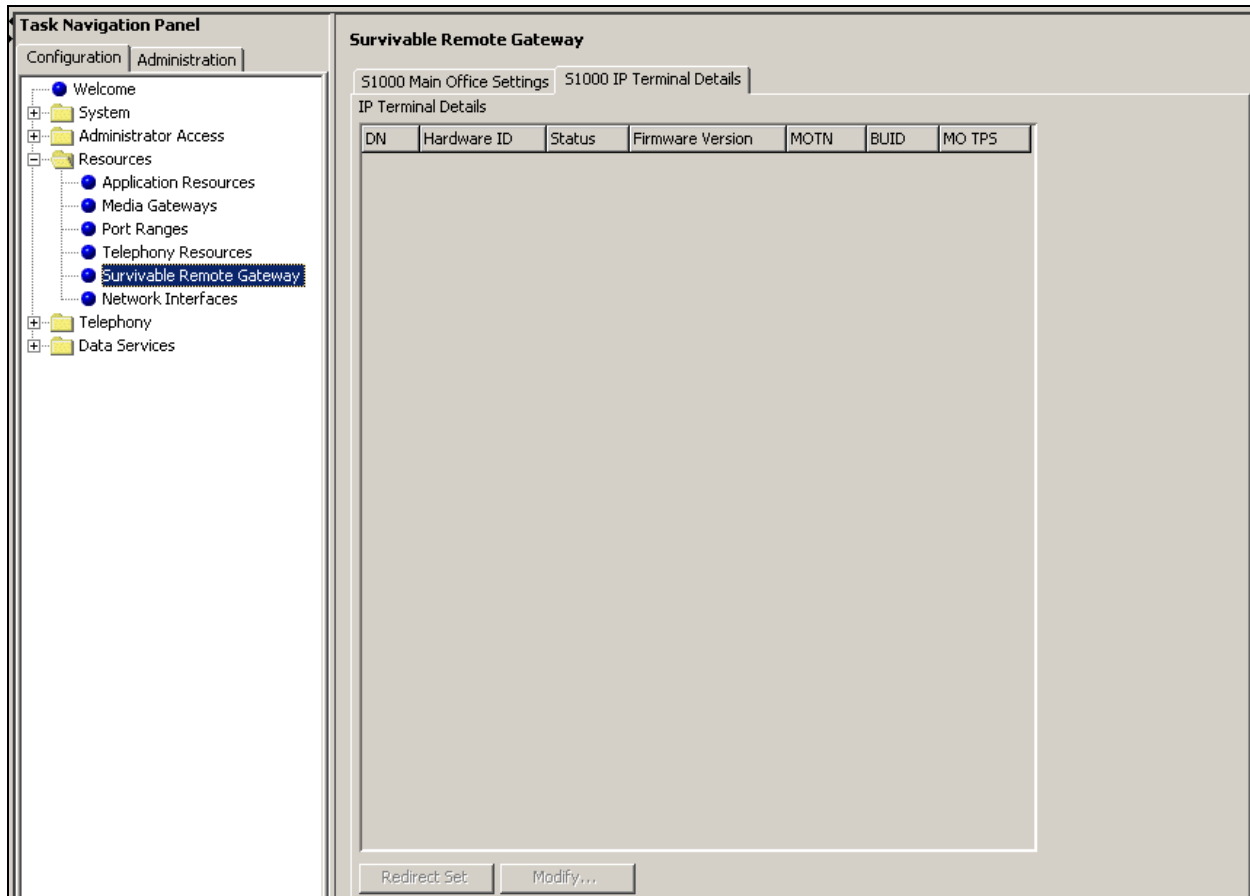
**Note:** The SRG DNs must be configured before the following procedures can be undertaken. See [Telephone \(DN\) records configuration](#) on page 56.

## To enter the MOTN and BUID

- 1 In Element Manager, navigate to **Configuration > Resources > Survivable Remote Gateway**.
- 2 Select the **S1000 IP Terminal Details** tab (see the figure [S1000 IP Terminal Details panel](#) on page 50).
- 3 Refer to the numbers and models recorded in the table [IP telephone numbers and models](#) on page 47.
- 4 Select the required DN.
- 5 Press the **Modify** button.
- 6 Enter the MOTN and the BUID in the appropriate fields.

## To redirect the telephone to the main office call server

- 1 In Element Manager, navigate to **Configuration > Resources > Survivable Remote Gateway**.
- 2 Select the **S1000 IP Terminal Details** tab (see the figure [S1000 IP Terminal Details panel](#) on page 50).
- 3 Select the DN of the telephone to be redirected.
- 4 Press the **Redirect Set** button.

**Figure 6** S1000 IP Terminal Details panel

## IP telephone settings

For IP telephones that are redirected to the main office call server, incorporate the settings shown in the table [Configuration settings for redirected IP Phones](#) on page 50. The *Device Configuration Guide* (NN40020-300) and the *Networking Configuration Guide* (NN40020-603) provide detailed instructions for configuring IP telephones.

**Table 11** Configuration settings for redirected IP Phones

Parameter	Setting
S1 IP	SRG IP address
S1 Port	7300
S1 Action	1
S1 Retry Count	1
S2 IP	SRG IP address
S2 Port	7300
S2 Action	1
S2 Retry Count	1

## Firmware upgrade

The redirected IP telephones at the SRG are under the control of the main office call server for the majority of their deployment and receive all of their features in that context. Therefore, the version of IP set firmware must align with the requirements of the CS 1000.

### Supported firmware

The table [Supported IP clients and firmware versions](#) on page 51 lists the IP clients and related firmware versions supported on the SRG50. The SRG50 column indicates the firmware versions included with the SRG software. The CS 1000 columns identify the version of firmware to use for specific releases.

**Table 12** Supported IP clients and firmware versions

IP Client	SRG50	CS 1000		
		Release 4.0	Release 4.5	Release 5.0
Phase I: 2002, 2004	B65 or greater	B65 or greater	B76 or greater	B76 or greater
Phase II: 2001, 2002, 2004	D98 or greater	D88 or greater	0604DBG or 0604DAS	0604DBG
IP Phone 2007	C22 or greater	C22 or greater	0621C4J or 0621C3J	0621C4J
WLAN Handsets 2210/2211	Not embedded in SRG software	Use the 2210/2211 GA firmware load.	97.059 or greater	97.059 or greater
WLAN Handset 2212	Not embedded in SRG software	97.039 or greater	97.059 or greater	97.059 or greater
WLAN Handsets 6140/6120	Not embedded in SRG software			
WLAN Handsets 1210/1220/1230	062AC57			
IP Phone 2033	Not embedded in SRG software	S12 or greater	S12 or greater	S12 or greater
IP Phone 1110	0623C38	0624C1E	0623C4B or 0623C3C	0623C4D
IP Phone 1120E	0624C1B	0624C1E	0624C4B or 0624C3C	0624C4D
IP Phone 1140E	0625C1B	0625C1E	0625C4B or 0625C3C	0625C4D
IP Softphone 2050	build 385 or greater	build 385 or greater	build 385 or greater	2.1.355 or greater
MVC 2050	build 126 or greater	build 126 or greater	build 126 or greater	2.1.202 or greater

### Firmware upgrade procedure

When an IP telephone requires a firmware upgrade, the CS 1000 uses the umsUpgradeAll command, or variant, to redirect the telephone back to the SRG for upgrading. If the required file does not exist on the SRG, or its version is incorrect, the SRG initiates an FTP session to the TPS for that phone to retrieve the required file. The SRG upgrades the phone and redirects it back to the CS 1000. After the SRG receives new firmware from the Main Office for a specific IP telephone, all the IP telephones of the same type are loaded with the new firmware.



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# Chapter 5

## IP telephones setup and configuration

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IP telephone setup and DN configuration are described in detail in the *IP Telephone Installation and Configuration Guide* and the *Networking Configuration Guide* (NN40020-603), respectively. SRG-specific procedures and settings include:

- [Registration password](#) on page 53
- [Local mode indication](#) on page 55
- [IP telephone codec and jitter settings](#) on page 55
- [Telephone \(DN\) records configuration](#) on page 56
- [Received numbers configuration](#) on page 59
- [DHCP settings configuration](#) on page 60
- [Call forwarding options](#) on page 60
- [Configuration settings for redirected phones](#) on page 61
- [Test Local Mode](#) on page 61
- [Features in local mode](#) on page 62
- [911 Emergency Services Support](#) on page 63

### Registration password

If a registration password is configured on the SRG, the IP telephone installer must enter the password before the telephone can be configured.

#### To set the IP telephone registration password

- 1 In Element Manager, navigate to **Configuration > Resources > Telephony Resources**.
- 2 On the **Modules** panel, locate the **Module type** column and select the **IP & Application Sets** row (see the figure [Telephony Resources panel, IP and App Sets](#) on page 54).
- 3 On the **Details for Module:** subpanel, select the **IP Terminal Global Settings** tab.

There are three fields that define the password registration process (see the table [Password registration parameters](#) on page 54):

**Enable registration, Enable global registration password, and Global password.**

**Table 13** Password registration parameters

<b>Enable registration</b>	Must be selected to allow IP telephones to register.
<b>Enable global registration password</b>	Select if a password is going to be entered in the next field, <b>Global password</b> .  If selected, the installer must enter the Global password (below) at the IP telephone before the telephone can be configured.
<b>Global password</b>	Enter the password.  If no password is entered, or the <b>Enable global registration password</b> is not selected, no password is required to configure the IP telephone.  <b>Note:</b> Nortel recommends that you synchronize this password with the CS 1000 password.

**Figure 7** Telephony Resources panel, IP and App Sets

The screenshot shows the 'Telephony Resources' panel. On the left is a 'Task Navigation Panel' with a tree view containing 'Welcome', 'System', 'Administrator Access', 'Resources', 'Application Resources', 'Media Gateways', 'Port Ranges', 'Telephony Resources' (selected), 'Dial Up Interfaces', 'Survivable Remote Gateway', 'Telephony', and 'Data Services'. The main area is titled 'Telephony Resources' and contains a table of modules:

Location	Module type	Bus	State	Devices	Low	High	Total	Busy
Internal	IP Sets	1	N/A	Sets	N/A	N/A	1	0
Internal	IP Trunks	N/A	N/A	Lines	1	24	24	0
Internal	BRI Loop	3	Enabled	Lines	61	64	4	0
Internal	Sets	4	Enabled	Sets	N/A	N/A	4	0
Expansion 1	Empty	5	N/A	N/A	N/A	N/A	N/A	N/A
Expansion 2	Empty	7	N/A	N/A	N/A	N/A	N/A	N/A

Below the table are 'Disable' and 'Enable' buttons. The 'Details for Module: Internal' section has two tabs: 'IP Terminal Global Settings' and 'IP Terminal Details'. A red arrow points to the 'IP Terminal Details' tab, which contains the following configuration options:

- Enable registration:
- Enable global registration password:
- Global password:
- Auto-assign DNS:
- Advertisement/Logo:
- Default codec:
- Default jitter buffer:
- G.729 payload size (ms):
- G.723 payload size (ms):
- G.711 payload size (ms):

- 4 Select and enter the values to meet the password requirements of your installation.
- 5 Set the **Auto-assign DNS** check box according to the requirements of your installation (if set, the SRG automatically assigns DNS, see the *Device Configuration Guide* (NN40020-300) for details).

## Local mode indication

When an IP telephone is in local mode, a message is displayed on the phone to indicate the local mode state to the user. The default setting is **Local Mode**.

### To change local mode indication

- 1 In Element Manager, navigate to **Configuration > Resources > Telephony Resources**.
- 2 On the **Modules** panel, locate the **Module type** column and select the **IP Sets** row (see the figure [Telephony Resources panel, IP and App Sets](#) on page 54).
- 3 On the **Details for Module** subpanel, select the **IP Terminal Global Settings** tab.
- 4 The **Advertisement/Logo** field specifies the message that provides local mode indication. Change as required.

## IP telephone codec and jitter settings

When the IP telephones are operating in local mode, codec and jitter settings are set on a phone-by-phone basis. Configure the settings to meet the requirements of the local SRG environment. They do not have to be the same as the main office settings (in contrast to the QoS settings for the VoIP trunks; see [QoS settings \(codec, jitter buffer, and related items\)](#) on page 70).

### To enter codec and jitter settings for IP telephones in local mode

- 1 In Element Manager, navigate to **Configuration > Resources > Telephony Resources**.
- 2 On the **Modules** panel, locate the **Module type** column and select the **IP Sets** row (see the figure [Modules panel, IP & Application Sets](#) on page 56).
- 3 On the **Details for Module** subpanel, select the **IP Terminal Global Settings** tab.  
The fields related to QoS are on the right side of the panel.
- 4 Enter the appropriate values.

Figure 8 Modules panel, IP &amp; Application Sets

**Task Navigation Panel**

Configuration | Administration

- Welcome
- System
- Administrator Access
- Resources
  - Application Resources
  - Media Gateways
  - Port Ranges
  - Telephony Resources**
  - Dial Up Interfaces
  - Survivable Remote Gateway
- Telephony
- Data Services

**Telephony Resources**

Modules

Location	Module type	Bus	State	Devices	Low	High	Total	Busy
Internal	IP Sets	1	N/A	Sets	N/A	N/A	1	0
Internal	IP Trunks	N/A	N/A	Lines	1	24	24	0
Internal	BRI Loop	3	Enabled	Lines	61	64	4	0
Internal	Sets	4	Enabled	Sets	N/A	N/A	4	0
Expansion 1	Empty	5	N/A	N/A	N/A	N/A	N/A	N/A
Expansion 2	Empty	7	N/A	N/A	N/A	N/A	N/A	N/A

Disable Enable

Details for Module: Internal

IP Terminal Global Settings | IP Terminal Details

Enable registration  Default codec: Auto

Enable global registration password  Default jitter buffer: Auto

Global password: \*\*\*\*\* G.729 payload size (ms): 30

Auto-assign DNS  G.723 payload size (ms): 30

Advertisement/Logo: Local Mode G.711 payload size (ms): 30

## Telephone (DN) records configuration

DN records for IP telephones are configured through the **All DN**s panel (**Configuration > Telephony > Sets > All DN**s) (see the figure [All DN](#)s panel on page 58). The *Networking Configuration Guide* (NN40020-603) and the *Device Configuration Guide* (NN40020-300) provide basic instructions for configuring DN

s and IP telephones. The following instructions are in addition to these instructions and only apply to IP telephones that are to be redirected to a main office call server.

**Note:** It is assumed that the line pools have been assigned. In the default configuration, VoIP trunks are assigned to line pool BlocA and the four PSTN trunks are assigned to line pool A. For more information, see [SRG50 overview](#) on page 15.

### To configure DN records for redirected IP telephones

- 1 In Element Manager, navigate to the **All DN**s panel (**Configuration > Telephony > Sets > All DN**s).
- 2 Select the **Line Access** tab.
- 3 Identify the row of the DN record to be configured.
- 4 Refer to the list of numbers and phone models recorded in the table (see [CS 1000 considerations](#) on page 33).



- 5 From the **Model** list, select the model of telephone assigned to this DN.  
For CS 1000: See [IP telephones numbers and models](#) on page 47.
- 6 To support outgoing caller ID over the VoIP trunk, the Private OLI field (**Priv. OLI**) must be set to the DN.
- 7 To support outgoing number display over the PSTN, enter the public access number for the telephone in the Public OLI field (**Pub. OLI**).
- 8 Leave the **Fwd All** field blank to disable the Call Forward All Calls feature when the telephone is in local mode. In normal mode, the SRG can forward all calls to the main office call server automatically (see [Call forwarding options](#) on page 60). In local mode, the Call Forward All Calls feature is automatically discontinued and the SRG routes calls to the SRG DNs.  

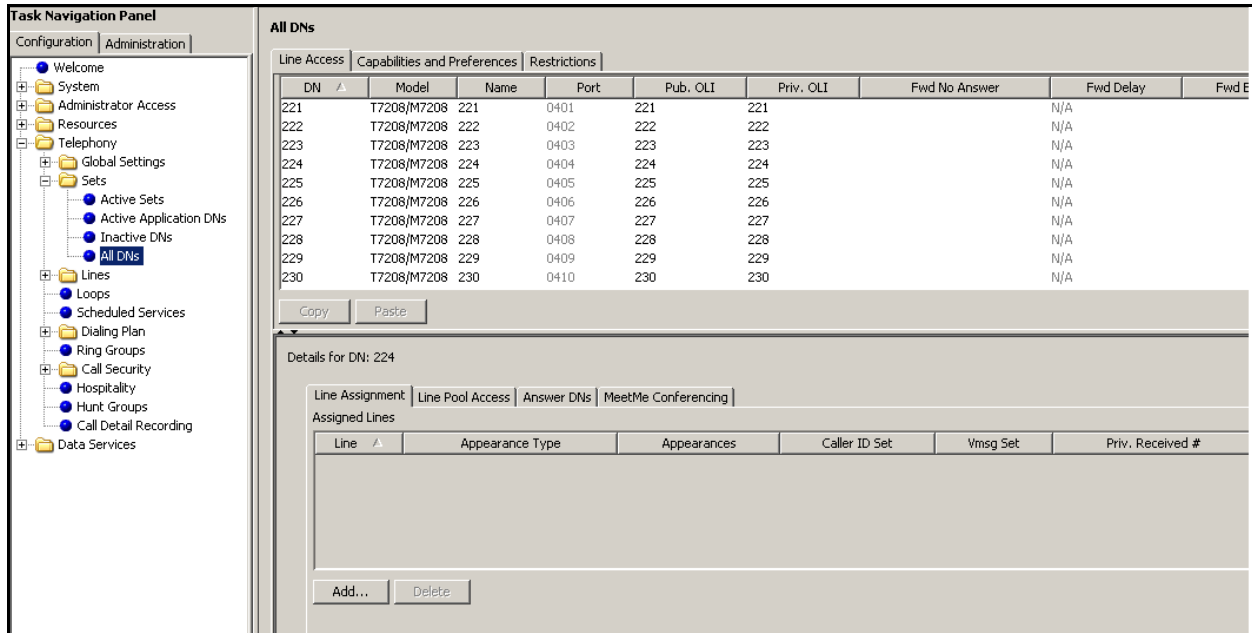
Call Forward All Calls does not require **Allow redirect** to be enabled (**Allow redirect** is found on the **Capabilities** tab of the **Details for DN** subpanel when the **Capabilities and Preferences** tab is selected).
- 9 To assign specific PSTN lines to each telephone, add the line(s) (**Details for DN** details panel > **Line Assignment** tab). You would do this if, for example, you want one user to field all customer calls when the system is in local mode (see the figure [All DNs panel](#) on page 58).
- 10 Ensure that the **Appearance Type** (**Details for DN** details panel > **Line Assignment** tab) is set to **Ring only**.
- 11 Assign the VoIP and PSTN trunk line pools to the DN (**Details for DN** details panel > **Line Pool Access** tab).
- 12 Assign a target line to the DN (**Details for DN** details panel > **Line Pool Access** tab). For a description of target lines, see [SRG50 terminology](#) on page 20.



**Note:** At this point, you may want to configure the received numbers. Refer to [Received numbers configuration](#) on page 59 and then return to this procedure.

---

**Figure 9** All DN's panel



- 13 Select the **Capabilities and Preferences** tab.
- 14 In the **Intercom keys** column, set **Intercom Keys** to 1 if required.
- 15 On the **Details for DN** subpanel, select the **Button Programming Table** tab.
- 16 Program the voice mail access button with the PSTN dialup for the main office voice message system.

Model	Button
2001	Message
2002	06
2004	08
2007	08
2033	08
2050, 2050CE	08
2210	08
2211	08
2212	08
1110	08
1120E	08
1140E	08

## Received numbers configuration

The **Public Received number length** and the **Private Received number length** (see [Basic parameters](#) on page 67) determine the number of digits that the SRG retains for call processing. The retained digits are mapped to the DN using fields provided on the **Target Lines** panel. (For more information on target lines, see [SRG50 terminology](#) on page 20).



**Note:** If the retained digits are the same as the DN, the fields (**Pub. Received #** field and **Priv. Received #** field) can be left blank.

### To map received numbers to the DN

- 1 In Element Manager, navigate to **Configuration > Telephony > Lines > Target Lines** (see the figure [Target Lines panel](#) on page 59).
- 2 Select the target line of the DN for which you want to configure the received numbers.
- 3 Double-click the **Pub. Received #** field, and then enter the retained received digits for calls originating from the PSTN.
- 4 Double-click the **Priv. Received #** field, and then enter the retained received digits for calls originating from the private network (that is, the VoIP trunks). This number is usually the same as the DN.

**Figure 10** Target Lines panel

The screenshot shows the 'Target Lines' panel in Element Manager. On the left is the 'Task Navigation Panel' with a tree view showing 'Configuration' > 'Administration' > 'Telephony' > 'Lines' > 'Target Lines' selected. The main area displays a table of target lines:

Line	Trunk Type	Name	Control Set	Line Type	Prime Set	Pub. Received #
125	Target line	Line125	221	Public	221	
126	Target line	Line126	221	Public	221	
127	Target line	Line127	221	Public	221	
128	Target line	Line128	221	Public	221	
129	Target line	Line129	221	Public	221	
130	Target line	Line130	221	Public	221	
131	Target line	Line131	221	Public	221	
132	Target line	Line132	221	Public	221	
133	Target line	Line133	221	Public	221	
134	Target line	Line134	221	Public	221	

Below the table are 'Copy' and 'Paste' buttons. The 'Details for Line: 126' section is visible, showing the 'Assigned DNs' tab. It includes fields for 'Aux. ringer' (checkbox), 'Distinct rings in use' (text box with 'None'), 'If Busy' (dropdown menu with 'To prime'), 'Voice message center' (text box with '1'), and 'Redirect to' (dropdown menu).

## DHCP settings configuration

### To configure DHCP settings for SRG operation

- 1 In Element Manager, navigate to **Configuration > Data Services > DHCP Server**.
- 2 Select the **General Settings** tab.
- 3 From the **DHCP Server is** list, select **Enabled - IP Phones Only**.
- 4 From the **WINS node type** list, select **H-node**.
- 5 In the Default gateway field, enter an address that meets the requirements of the SRG50 LAN.
- 6 In the Lease time field, enter a value that meets the requirements of your system.  
Leave all other fields under the General Settings tab blank.
- 7 Select the **IP Terminal DHCP Options** tab.
- 8 On the **Primary Terminal Proxy Server (S1)** and **Secondary Terminal Proxy Server (S2)** subpanels:
  - a In the **IP address** field, enter the IP address of the SRG.
  - b From the **Port** list, select **SRG**.
  - c In the **Retry count** field, enter the number of retries that the IP telephone is allowed to connect to the SRG before an event is generated (see [IP terminal details](#) on page 84).
- 9 Select the **Address Ranges** tab and add a range of IP address to meet the requirements of your system.

## Call forwarding options

There are two options for configuring call forwarding on the SRG:

- The target DN is determined by the BUID/MODN and VoIP Trunk access codes.

In this case, the information required for call forwarding is entered using the SRG-specific panels of Element Manager (**Configuration > Resources > Survivable Remote Gateway**). For more information, see [CS 1000 considerations](#) on page 33.



**Note:** Call forwarding is mandatory for CS 1000.

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- The target DN is configured explicitly for each IP telephone.

The DN is configured through Element Manager on the **Telephony > Sets > Active Sets** panel using the standard BCM50 procedure. Typically, the call forward number is the BUID. For more details, see the *Device Configuration Guide* (NN40020-300).

The disadvantage of the second option is that the installer must configure the target DN in two places: on the **Active Sets** panel and on the SRG-specific panels (the SRG-specific panels **must** be completed). The SRG looks at the SRG-specific panels first. It goes to the **Active Sets** panel only if the **VOIP Trunk Access Code** has not been configured.

## Configuration settings for redirected phones

The *Telephony Device Installation Guide* and the *Networking Configuration Guide* provide detailed instructions for configuring IP telephones. For IP telephones that are redirected to the main office call server, incorporate the settings shown in the table [Configuration settings for redirected IP telephones](#) on page 61.

**Table 14** Configuration settings for redirected IP telephones


Parameter	CS 1000
S1 IP	SRG IP address
S1 Port	7300
S1 Action	1
S1 Retry Count	1
S2 IP	SRG IP address
S2 Port	7300
S2 Action	1
S2 Retry Count	1

For more information, see [CS 1000 considerations](#) on page 33.

## Test Local Mode


An IP telephone operating in normal mode can be forced to redirect to the SRG. This allows the telephone user, and system administrator, to test local mode operation without taking down the VoIP trunk to the main office.

### To activate Test Local Mode

- 1 From the telephone dialpad, press the **Services** key  .
- 2 Press the **Test Local Mode** button.

The telephone is now in Test Local Mode.

## To exit Test Local Mode (and return to Normal Mode)

Generally, you exit Test Local Mode by waiting for the feature to time out or by pressing the Exit key (  ). This key is active only when the telephone is in the local mode test.

For the WLAN Handsets 2210/2211/2212, pressing the End key causes the phone to exit Test Local Mode.

If the phone does not have an Exit key, you must wait until the test times out. For more information about Test Local Mode Timeout, see [S1000 Main Office Settings panel](#) on page 44.

## Features in local mode

In local mode, IP telephones at the SRG no longer have access to the full suite of main office applications. However, the SRG does provide a set of features that include connectivity with the local PSTN, access to Emergency Services, and the ability to call local extensions.

The SRG also supports the following features in local mode:

- Hold
- Transfer (dedicated key on the 2002, 2004, and 2007 models)
- Call Forward No Answer/Busy (if the feature has been enabled on the DN: **Configuration > Telephony > Sets > All DNs**)
- Last Number Redial (dedicated key on the 2002/2004 models)
- Inbox Key (on 2002/2004 models)

The user experience in local mode can be enhanced if certain global feature settings are coordinated with the main office so that the settings are the same at both the main office and the SRG. These feature settings are configured with the **Feature Settings** panel on the Element Manager interface (**Configuration > Telephony > Global settings > Feature Settings**). Feature settings that can be coordinated with the main office are:

- **Background music** (if it is provided for on-hold)
- **On-hold**  
This determines if a caller on hold hears tones, music, or nothing.
- **Receiver volume**  
Set to use the system volume, since IP users cannot use the feature code to set a default telephone volume.
- **Delayed ring transfer**  
If a transfer to an external number is not answered, you can indicate if the call will be dropped (Off) or transferred to the designated Prime telephone.

Check the **Transfer callback timeout**. This setting defaults to **After 4 rings**. If you are using the **Delayed ring transfer** feature, turn **Transfer callback timeout** off if you want all unanswered transferred calls to ring at the Prime set (usually the system attendant). If you want the transferred call to ring at the telephone from which it was transferred first, set this field to a setting that is less than the setting for **Delayed ring transfer**.

- **Held line reminder**  
If set to a time, determines period between when a call is put on hold and when a short tone sounds at the telephone to indicate the call is still on hold.
- **Alarm set**  
The alarm set feature is not supported on SRG.
- **Language and Contrast**  
Language and Contrast are DN-specific settings and are configured at **Configuration > Telephony > Sets > All DNs > All DNs panel > Capabilities and Preferences tab > Details for DN subpanel > Preferences tab**.

Features not supported in local mode include: Hot Desking, Do Not Disturb, Page, Call Forward, Background Music, Call Park, Call Pickup, Speed Dial, and Conference.

## 911 Emergency Services Support

For IP telephones in local mode, and for other telephones at the SRG, the *Networking Configuration Guide* (NN40020-603) provides details for configuration of 911 emergency services.

For redirected IP telephones in normal mode, the IP telephone is registered with the main office call server. Ensure that the main office call server is configured so that a 911 call from an IP telephone at the SRG is routed back to the SRG's local PSTN. [Emergency Services Access \(ESA\) configuration](#) on page 42 includes a procedure for configuring the SRG for CS 1000 Emergency Services Access.





## Chapter 6

# Setting up the private VoIP network

To provide SRG functionality and to take advantage of VoIP technology, a private VoIP network is required between the SRG and the main office. This chapter details the procedures for establishing appropriate WAN connections to enable a VoIP network between the main office and SRG branch locations. Before proceeding, ensure that IP networking from the SRG to the WAN, and from the main office call server to the WAN have been configured and tested (see the figure [IP networking, SRG to WAN, main office to WAN](#) on page 65). SRG-specific configuration establishes the VoIP network (see the figure [VoIP path for SRG operation](#) on page 66).

**Figure 11** IP networking, SRG to WAN, main office to WAN

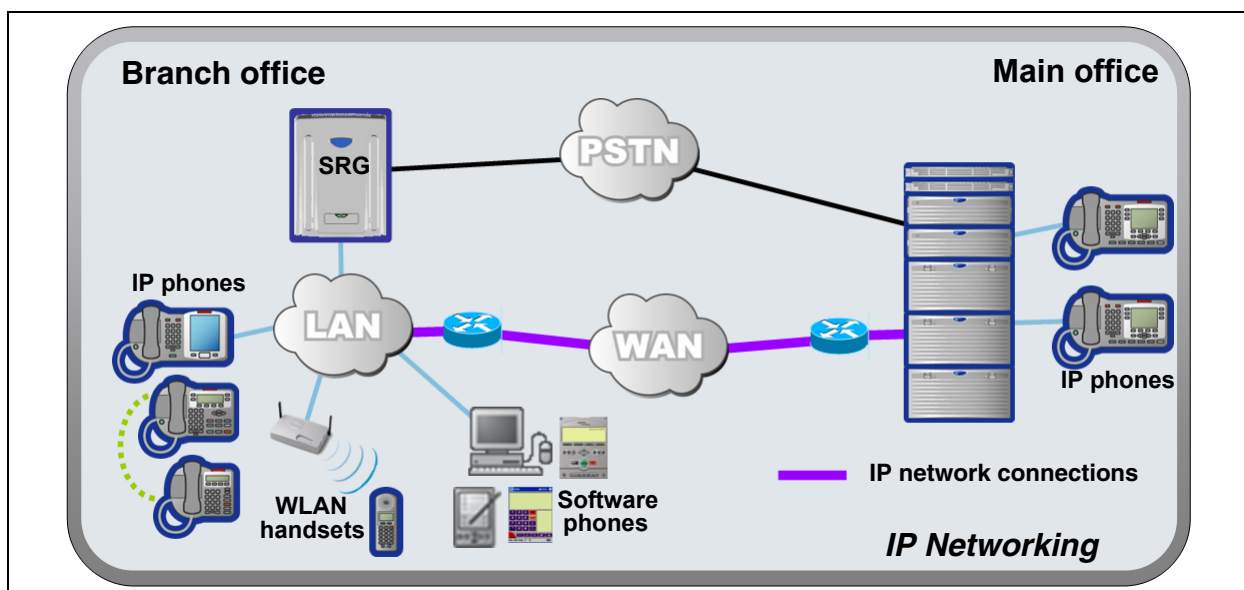
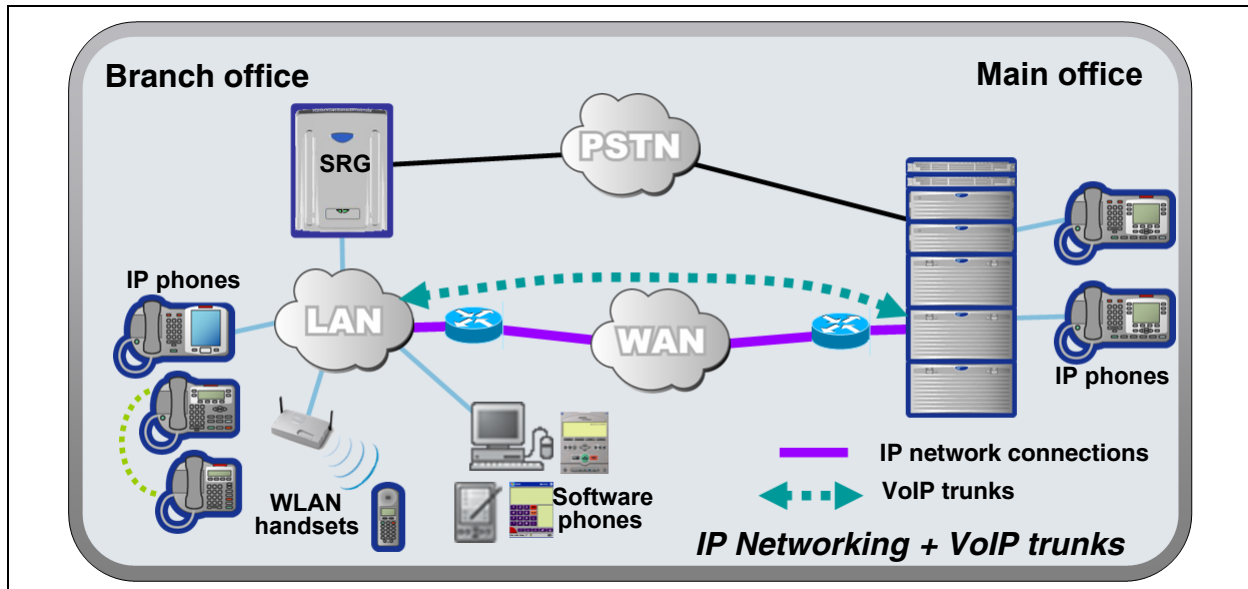


Figure 12 VoIP path for SRG operation



Generic procedures for setting up a private network on the SRG are covered in the *Networking Configuration Guide* (NN40020-603). Items to address when establishing the private VoIP network between the SRG and the main office are:

- [Basic parameters](#) on page 67
- [Private dialing plan](#) on page 68
- [Meridian Customer Defined Network \(MCDN\)](#) on page 69
- [QoS settings \(codec, jitter buffer, and related items\)](#) on page 70
- [Network security](#) on page 72
- [VoIP trunking configuration](#) on page 72
- [Line pools](#) on page 74
- [Call routing](#) on page 74
- [Outgoing calls configuration](#) on page 74
- [SRG PSTN access](#) on page 76
- [Main office information](#) on page 78
- [External attendant support](#) on page 78

## Basic parameters

The table [Basic parameters](#) on page 67 provides a record of basic parameters that are significant for SRG operation. Typically, these parameters are specified as part of BCM50 foundation activities; in most cases, their configuration is not covered in the SRG50 Configuration Guide.

**Table 15** Basic parameters (Sheet 1 of 2)

Parameter	Value	Context
DN length		<p>Configured as part of BCM50 foundation configuration.</p> <p>There are four DN lengths to consider.</p> <ol style="list-style-type: none"> <li>1. Configuration &gt; Telephony &gt; Dialing Plan &gt; General &gt; Dialing Plan - General panel &gt; Global Settings subpanel &gt; DN length (intercom) field This is the internal DN length. That is, the length of DNs for calls between telephones on the SRG.</li> <li>2. Configuration &gt; Telephony &gt; Dialing Plan &gt; Public Network &gt; Dialing Plan - Public Network panel &gt; Public Network Settings subpanel &gt; Public Received number length field For calls originating from the PSTN, this establishes the number of digits the SRG retains.</li> <li>3. Configuration &gt; Telephony &gt; Dialing Plan &gt; Private Network &gt; Dialing Plan - Private Network panel &gt; Private Network Settings subpanel &gt; Private Received number length field For calls originating from a private network, this establishes the number of digits the SRG retains.</li> <li>4. Configuration &gt; Telephony &gt; Dialing Plan &gt; Private Network &gt; Dialing Plan - Private Network panel &gt; Private Network Settings subpanel &gt; Private DN length field Used for DPNSS applications only. See the <i>Networking Configuration Guide</i> (NN40020-603).</li> </ol>
DN range		<p>Configured as part of BCM50 foundation configuration.</p> <p>DN range of the SRG is set by the hardware configuration and keycodes. Actual numbering is contiguous from the Start DN: Administration &gt; Utilities &gt; Reset &gt; Reset panel &gt; Cold Reset Telephony Services button &gt; Cold Reset Telephony dialog box &gt; Start DN field</p>
Destination code for VoIP trunks		<p>Configured for advanced routing. See <a href="#">Outgoing calls configuration</a> on page 74</p> <p>VoIP destination code = VOIP Trunk Access Code* = AC1**</p> <p>* VOIP Trunk Access Code is entered on the main office settings panel. See the server-specific chapters for details.</p> <p>** For a UDP dialing plan, AC1 is the access code in the digit string &lt;AC1&gt; &lt;LOC&gt; &lt;DN&gt;.</p>
SRG IP address		<p>Configured as part of BCM50 foundation configuration.</p> <p>Configuration &gt; System &gt; IP Subsystem &gt; IP Subsystem panel &gt; General Settings tab &gt; IP Settings details subpanel</p>

**Table 15** Basic parameters (Sheet 2 of 2)

Parameter	Value	Context
SRG net mask		Configured as part of BCM50 foundation configuration. Configuration > System > IP Subsystem > IP Subsystem panel > General Settings tab > IP Settings details subpanel
IP address of SRG gateway		Configured as part of BCM50 foundation configuration. Configuration > System > IP Subsystem > IP Subsystem panel > General Settings tab > IP Settings details subpanel
VLAN		If the SRG operates as part of a VLAN, obtain the required identifiers from the VLAN administrator. Configuration > Data Services > DHCP Server > DHCP Server panel > IP Terminal DHCP Options tab > VLAN Identifiers subpanel
PSTN number for dialing into the main office from the SRG when in local mode		Required to specify a PSTN fallback route. See <a href="#">Outgoing calls configuration</a> on page 74.
PSTN number for dialing into the SRG from the main office when in local mode		Required for main office configuration.

## Private dialing plan

For SRG operation, either a coordinated dialing plan (CDP) or a uniform dialing plan (UDP) can be configured. Nortel recommends CDP because it requires the least dialing manipulation between the SRG and the main office call server. The dialing plan choice also determines whether the DN on the SRG matches the BUID.

Dialing plans between the SRG and the main office call server must be compatible. Private dialing plan configuration is described in detail in the *Networking Configuration Guide* (NN40020-603).

The type of dialing plan, CDP or UDP, is determined by the main office configuration.

The path to the Element Manager panel for setting up the dialing plan is Configuration > Telephony > Dialing Plan > Private Network > Dialing Plan - Private Network panel > Private Network Settings subpanel (see the figure [Dialing Plan — Private Network panel, Private Network Settings subpanel](#) on page 69).

**Figure 13** Dialing Plan — Private Network panel, Private Network Settings subpanel

**Task Navigation Panel**

Configuration Administration

- Welcome
- System
- Administrator Access
- Resources
- Telephony
  - Global Settings
  - Sets
  - Lines
  - Loops
  - Scheduled Services
  - Dialing Plan
    - General
    - DNs
    - Public Network
    - Private Network
    - Line Pools
    - Routing
    - Ring Groups
  - Call Security
    - Hospitality
    - Hunt Groups
    - Call Detail Recording
  - Data Services
  - Applications

**Dialing Plan - Private Network**

Private Network Settings

Private Received number length: 3

Private Auto DN:

Private DISA DN:

Private access code:

Private network type: None

Location code:

Private DN length: 3

**MCDN**

Local access code:

National access code:

Special access code:

Network ICCL:

TRO:

TAT:

**VoIP**

Virtual Private Network ID: 0

Zone ID: 0

**ETSI**

Network Diversion:

MCID:

## Meridian Customer Defined Network (MCDN)

MCDN is automatically activated when the system is converted to SRG operation. To ensure that redirected IP telephones can transfer calls to the SRG local telephones, trunk anti-tromboning (TAT) must be enabled.

### To enable MCDN TAT

Use Element Manager to enable TAT by selecting the TAT checkbox at Configuration > Telephony > Dialing Plan > Private Network > Dialing Plan - Private Network panel > MCDN subpanel (see the figure [Dialing Plan — Private Network panel, MCDN subpanel](#) on page 70).

**Figure 14** Dialing Plan — Private Network panel, MCDN subpanel

**Task Navigation Panel**

Configuration Administration

- Welcome
- System
- Administrator Access
- Resources
- Telephony
  - Global Settings
  - Sets
  - Lines
  - Loops
  - Scheduled Services
  - Dialing Plan
    - General
    - DNs
    - Public Network
    - Private Network
    - Line Pools
    - Routing
  - Ring Groups
- Call Security
  - Hospitality
  - Hunt Groups
  - Call Detail Recording
- Data Services
- Applications

**Dialing Plan - Private Network**

Private Network Settings

Private Received number length: 3

Private Auto DN:

Private network type: None

Private DISA DN:

Location code:

Private access code:

Private DN length: 3

MCDN

Local access code:

National access code:

Special access code:

Network ICCL:

TRO:

TAT:

VoIP

Virtual Private Network ID: 0

Zone ID: 0

ETSI

Network Diversion:

MCID:

## QoS settings (codec, jitter buffer, and related items)

Quality of Service (QoS) settings for the VoIP trunks at the SRG are determined by the main office settings; the SRG settings must match the main office. Use the table [Main office QoS settings](#) on page 70 to record the main office settings, to facilitate configuration, and to provide a record of the datafill.

**Table 16** Main office QoS settings (Sheet 1 of 2)

Media parameter	H.323 settings	SIP settings
Enable Voice Activity Detection		
Codec Preferences		
Silence compression (yes/no)		--
Jitter buffer		
G.729 payload size (ms)		

**Table 16** Main office QoS settings (Sheet 2 of 2)

Media parameter	H.323 settings	SIP settings
G.723 payload size (ms)	Supported for IP telephones for local calls only. See <a href="#">IP telephone codec and jitter settings</a> on page 55 for details.	
G.711 payload size (ms)		
Incremental payload size (yes/no)		--
T.38 fax (yes/no)		--
Force G.711 for 3.1k audio (yes/no)		--
Fax Transport	--	
Provide in-band ringback	--	

In Element Manager, enter QoS settings for VoIP trunks through the Telephony Resources panel.

## To enter the QoS settings for VoIP trunks

- 1 In Element Manager, navigate to **Configuration > Resources > Telephony Resources**.
- 2 On the **Modules** panel, locate the **Module type** column and select the **IP Trunks** row (see the figure [Modules panel, IP Trunks](#) on page 71).
- 3 On the **Details for Module** subpanel, select the **H323 Media Parameters** tab or the **SIP Media Parameters** tab depending on which media parameters you want to configure.
- 4 Refer to the main office QoS settings recorded in the table [Main office QoS settings](#) on page 70 and enter the appropriate values.

**Figure 15** Modules panel, IP Trunks

The screenshot displays the 'Telephony Resources' configuration page. On the left is the 'Task Navigation Panel' with a tree view containing 'System', 'Administrator Access', 'Resources', 'Application Resources', 'Media Gateways', 'Port Ranges', 'Telephony Resources' (highlighted), 'Dial Up Interfaces', 'Survivable Remote Gateway', 'Telephony', and 'Data Services'. The main area shows a table of modules:

Location	Module type	Bus	State	Devices	Low	High	Total	Busy
Internal	IP Sets	1	N/A	Sets	N/A	N/A	1	0
Internal	IP Trunks	N/A	N/A	Lines	1	24	24	0
Internal	BRI Loop	3	Enabled	Lines	61	64	4	0
Internal	Sets	4	Enabled	Sets	N/A	N/A	4	0
Expansion 1	Empty	5	N/A	N/A	N/A	N/A	N/A	N/A
Expansion 2	Empty	7	N/A	N/A	N/A	N/A	N/A	N/A

Below the table are 'Disable' and 'Enable' buttons. The 'Details for Module: Internal' subpanel is open, showing tabs for 'Routing Table', 'IP Trunk Settings', 'H323 Settings', 'H323 Media Parameters', 'SIP Settings', 'Sip Proxy', 'SIP Media Parameters', 'SIP URI Map', and 'SIP Authentication'. The 'SIP Settings' tab is active, displaying a table with columns: 'Description', 'Destination Digits', 'Domain', 'IP Address', 'Port', 'GW Type', and 'MCDN Protocol'. 'Add...' and 'Delete' buttons are at the bottom.

## Network security

Firewall configuration for SRG is the same as for the BCM50 and is detailed in the *Networking Configuration Guide* (NN40020-603). Firewalls cannot be configured to allow VoIP pass through. Instead, the SRG supports IPsec tunnels to provide VoIP pass through. IPsec tunnels are also covered in the *Networking Configuration Guide* (NN40020-603).

## VoIP trunking configuration

When the SRG is operating in normal mode, connectivity to the main office call server is over VoIP trunks. An SRG can support up to 60 VoIP trunks.

Configuring VoIP trunks has three components:

- [Fallback configuration](#) on page 72
- [Gatekeeper routing](#) on page 73
- [Line pools](#) on page 74

## Fallback configuration

For SRG operation, fallback and gatekeeper configuration is common to all VoIP trunks. Fallback is determined in conjunction with the QoS settings (see [To enter the QoS settings for VoIP trunks](#) on page 71).

**Note:** For calls originating from IP Sets, configure the payload size. Access the IP Sets configuration panel in Element Manager (**Configuration > Resources > IP Sets > IP Terminal Global Settings**). Change the **Payload Size** to 20ms.

## To enable fallback

- 1 Access the IP Trunks configuration panel in Element Manager (**Configuration > Resources > Telephony Resources > Modules** panel > **Module type** column) and then select **IP Trunks**.
- 2 On the Details for Module panel, select the **H323 Settings** tab or the **SIP Settings** tab.
- 3 On the Telephony Settings subpanel, locate the **Fallback to circuit-switched** field and use the list to:
  - a select **Enabled-All** if you want calls to be able to fallback to PSTN trunks if connectivity to the main office is lost.
  - b select **Enabled-TDM** if you want all TDM calls to be able to fallback to PSTN trunks if connectivity to the main office is lost.
  - c select **Disabled** if fallback is not required.



## Gatekeeper routing

The gatekeeper routes the calls based on an internal numbering table. Ensure that the gatekeeper administrator has a list of the numbers that identify the SRG and the SRG PSTN.

Examples:

- If the system is running with a CDP dialing plan and the SRG DN range is from 3000 to 3199, the gatekeeper requires this information to establish that calls starting with 30 and 31 are routed to the SRG.
- If the PSTN to which the SRG connects has a location code of 521, the gatekeeper must have a record of this code in the SRG list so that main office calls to the SRG PSTN can be routed correctly.

If you require H323 trunks, you must select the Gatekeeper wildcard check box in the H323 settings. If you do not select the Gatekeeper wildcard check box, then calls are SIP routed (default setting).

When you select the Gatekeeper wildcard check box, all dialed digits match gatekeeper digits and VoIP calls are routed through the gatekeeper.

For detailed information about H323 gatekeeper and SIP proxy, see the *Networking Configuration Guide* (NN40020-603).

## To configure gatekeeper settings for H323 trunks

- 1 Access the IP Trunks configuration panel in Element Manager (Configuration > Resources > Telephony Resources > Modules panel > Module type column) select **IP Trunks**.
- 2 On the Details for Module panel, select the **H323 Settings** tab.
- 3 On the Configuration subpanel, click **Modify**.
- 4 From the Call signaling list, select **Gatekeeper Resolved** or **Gatekeeper Routed**.
- 5 In the Primary Gatekeeper IP field, enter the IP address for the Primary Network Routing Services Address\*
- 6 In the Backup Gatekeeper(s) field, enter the IP address for Alternate Network Routing Services Address
- 7 In the Alias names field, enter **Name:** followed by the H.323 ID of the SRG (for naming conventions, see the *Networking Configuration Guide* (NN40020-603)).
- 8 Click **Ok**.
- 9 On the Telephony Setting subpanel, from the MCDN protocol list, select **CSE**.
- 10 On the Telephony Settings subpanel, select the **Gatekeeper wildcard** check box.

## SIP configuration

Access the IP Trunks configuration panel in Element Manager to configure the SIP trunks (Configuration > Resources > Telephony Resources > Modules panel > Module type column > IP Trunks, then select the SIP Settings tab).

When you set the value of the RFC2833 Dynamic Payload field, if the value is greater than 0, then this feature is enabled. If the feature is enabled then DTMF digits are sent in RTP packets. For CS1000 Release 4.5 set the value to 0.

For detailed information on entering the values in this panel, see the *Networking Configuration Guide* (NN40020-603).

## Line pools

Both VoIP trunks and PSTN trunks must be configured in separate line pools. In the default state, all VoIP trunks are assigned to line pool BlocA and all PSTN trunks are assigned to line pool A. It is not necessary to reassign the line pools.

Instructions for configuring line pools is provided in the *Networking Configuration Guide* (NN40020-603).

## Call routing

Call routing is covered in depth in the *Networking Configuration Guide*. The instructions in the SRG50 Configuration Guide are an abbreviation of that material, and only cover procedures that are specific to SRG operations; that is, for calls from redirected IP telephones. For more detailed information, see the *Networking Configuration Guide*.



**Note:** The DNs for the main office telephones system are marked off by the vacant number routing feature. SRG does not support Vacant Number Routing (VNR).

Instead, SRG uses Call Forward All Calls to emulate VNR for the SRG IP telephones that are in normal mode. When the telephones switch to local mode, Call Forward All Calls is cancelled for those telephones.

A single destination code and route (or a group of destination codes and routes) can be set up on the SRG to route all the calls that are not terminated locally by the SRG. These calls are routed over the VOIP trunks. In the case where the VoIP trunks become unavailable, the calls can be routed to the proper location using PSTN fallback. This is similar to the VNR feature in CS 1000.

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## Outgoing calls configuration

### To configure routing for outgoing calls

- 1 Create a schedule.
  - a Access the **Scheduled Services** panel (Configuration > Telephony > Scheduled Services).
  - b Select a **Schedule** (Sched 4, for example).

- c** Change the schedule name (optional).  
In this procedure, the name **SRG** is used as the name of the schedule.
    - d** In the Services subpanel, select your schedule.
    - e** In the Routing Svc row for your schedule, select **Auto** from the list.
    - f** Select the **Overflow** checkbox for your schedule.
    - g** Change the schedule time so that the schedule runs continuously (Start Time 00:00:00, Stop Time 23:59:59, MTWTFSS).
- 2** Define a route for calls to the main office over the VoIP trunks and a route to the main office over the PSTN.
  - a** Access the **Dialing Plan - Routing** panel (Configuration > Telephony > Dialing Plan > Routing).
  - b** Select the **Routes** tab.
  - c** Click **Add** to add a new route.
  - d** In the Route field, enter a number for the new route (for example, 998).
  - e** Click **Ok**.
  - f** Ensure that the DN Type is Public (Unknown).
  - g** Assign the VoIP line pool to the route (select the line pool from the **Use Pool** list; default is A).
  - h** Click **Add** to add another new route.
  - i** In the Route field, enter a number for the new route (for example, 999).
  - j** Click **Ok**.
  - k** Ensure that the DN Type is Public (Unknown).
  - l** In the External Number field, enter the **PSTN number** of the main office.
  - m** Assign the PSTN line pool to the route (select the line pool from the **Use Pool** list; default is A).
- 3** Add a destination code to provide access to the newly created routes. This code is used in both normal and local modes for dialing the main office from the SRG site.
  - a** Access the **Dialing Plan - Routing** panel (Configuration > Telephony > Dialing Plan > Routing).
  - b** Select the **Destination Codes** tab.
  - c** Click **Add** to add a new destination code.
  - d** In the Destination Code field, enter the new destination code (for example, 678).
  - e** Click **Ok**.
  - f** With the Destination Code row highlighted, select the **SRG** schedule from the Alternate Routes list (Alternate Routes for Destination Code details panel).
  - g** In the First Route field, enter **998** (the VoIP route).

- h** In the adjacent Absorbed Length field, select the number of digits to be absorbed. The Absorbed Length applies to the digits of the destination code only.
- i** In the Second Route field, enter **999** (the PSTN route).
- j** In the adjacent Absorbed Length field, select the number of digits to be absorbed.

Depending on the dialing plan, the destination code is integrated with the DN or is dialed as a prefix to the DN. When a user calls the main office, the SRG examines the destination code to determine the routing. If the First Route, the VoIP trunks, is unavailable, the call is routed to the Second Route, the PSTN, and the External Number is called. Because Overflow was selected, if both the First Route and the Second Route are unavailable, the call is routed using the Normal Route specified in the Normal Route column of the Destination Codes table. Because Auto was selected, the routing occurs without manual intervention.

## SRG PSTN access

Access to the SRG PSTN is required for:

- calls to the SRG PSTN from SRG telephones or redirected SRG IP telephones in local mode
- calls from the SRG PSTN to redirected SRG IP telephones in normal mode
- calls from main office telephones to the SRG PSTN, using the VoIP trunks

To achieve this access, a remote access package for the VoIP trunks and a destination code for the PSTN must be configured.

## Remote Access Package for VoIP trunks

The SRG views all calls coming in over the VoIP trunks as remote access calls, even though the VoIP pathway is a dedicated trunk to another closed system.

To allow tandem dialing from the main office through the SRG PSTN, or to redirect SRG IP telephones to use the SRG local PSTN, you must specify a remote package that provides access to the PSTN line pool. This remote package is then assigned to each VoIP trunk.

## To configure remote access packages

- 1** LocateSet up a remote access package for the PSTN line pool (Configuration > Telephony > Call Security > Remote Access Packages).
- 2** Assign the package to each VoIP trunk (Configuration > Telephony > Lines > Active VoIP Lines > Trunk Type column > Details for Line panel > Restrictions tab > Use remote package field).

## PSTN destination codes configuration

To allow SRG telephones to dial out over the PSTN and to allow main office telephones to tandem out through the local SRG PSTN, you need to define a destination code that accesses the PSTN line pool without an External Number. Frequently, this code is 9, but it does not have to be.

The following procedure provides a basic PSTN routing setup.

### To configure destination codes for the PSTN

- 1 Access the Dialing Plan - Routing panel (Configuration > Telephony > Dialing Plan > Routing).
- 2 Select the **Routes** tab. See the *Networking Configuration Guide* (NN40020-603) to:
  - a Add a new route (for the PSTN line pool).
  - b From the DN Type list, select **Public**.
  - c Leave the **External Number** field blank.
  - d Assign the PSTN line pool to the route (select the line pool from the **Use Pool** list).
- 3 Access the Dialing Plan - Routing panel (Configuration > Telephony > Dialing Plan > Routing).
- 4 Select the **Destination Codes** tab. See the *Networking Configuration Guide* (NN40020-603) to:
  - a Add the destination code to be used to access the local (SRG) PSTN.

Users on both SRG and main office telephones dial this destination code to access the local (SRG) PSTN. If this code goes only to the SRG PSTN, enter 9 + Wild Card 1. This wild card allows any numbers not used by other 9-based destination codes.



The default line pool access code for pool A is 9. Delete this access code before you attempt to create a destination code with 9.

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In normal mode, the destination code is forwarded from the main office to the SRG for SRG IP telephone calls that are connecting to the SRG PSTN.



**Note:** For main office programming, this code is the offnet dialing code that the gatekeeper recognizes for routing to the SRG.

At the main office, zone-based digit manipulation is used to add a Zone Digit Prefix (ZDP) to PSTN calls from SRG IP telephones. The ZDP allows the main office to differentiate between local PSTN calls made from main office telephones (to the main office PSTN) and PSTN calls made from SRG IP telephones (to the SRG PSTN). The main office administrator supplies this ZDP with the prerequisite information.

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- b Assign the Normal and SRG scheduled route for the two destination codes.

## Main office information

The SRG requires information about the main office call server that is not needed for a BCM50. Element Manager accommodates this information with SRG-specific panels that are activated after the SRG50 keycode is applied. The information required for these panels is specific to the main office call server [CS 1000 considerations](#) on page 33.

## External attendant support

The SRG can use the BCM50 Selective Line Redirection capability to provide an external attendant. If the attendant is located in the main office, there are two ways to maintain the attendant if the VoIP trunks become unavailable:

- 1 A fallback (or Prime) DN at the SRG can be specified. Since this DN is likely to receive calls in a WAN failure scenario, it must be an IP telephone that can transfer the calls to the desired party. If the IP set is also a redirected IP set, there is a period of time where inbound calls are un-routable, until the IP set falls back to the SRG.
- 2 A fallback route to the main office call server over the PSTN can be configured. At the main office, vacant number handling (such as routing to voice mail) can be applied.



**Note:** The SRG does not have local attendant capability.

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# Chapter 7

## PSTN access and analog devices

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SRG-specific items relevant to PSTN trunks and analog devices include:

- [PSTN access considerations](#) on page 79
- [Analog devices considerations](#) on page 79

### PSTN access considerations

Consider the following for PSTN access:

- **PSTN access**  
To provide access to the SRG PSTN when the SRG is in local mode, or to be able to set up tandem dialing from the main office through the SRG to the SRG PSTN, one or more PSTN trunks must be configured on the system. See the *Networking Configuration Guide* (NN40020-603).
- **Tandem calls**  
The SRG considers all calls coming in over the VoIP trunks as remote access calls, even though the VoIP pathway is a dedicated trunk to another closed system.  
  
To allow tandem dialing from the main office to the SRG PSTN, or to allow redirected SRG IP telephones to use the SRG local PSTN, a remote access package must be specified to provide access to the PSTN line pool. This procedure is covered in [Outgoing calls configuration](#) on page 74.
- **Manual- and auto-answer lines**  
If the trunk is configured as a manual-answer line:  
Enter the line pool access code and the dial string for the main office attendant telephone in the Redirect to field (Configuration > Telephony > Lines > Active Physical Lines).  
  
If the line is an auto-answer line in normal mode, incoming call requests are automatically call forwarded to the main office. When the SRG IP telephones revert to local mode, the system discontinues Call Forward All Calls and calls are delivered directly to the SRG IP telephones at the SRG.

### Analog devices considerations

Consider the following for analog devices:

- **Basic operation**  
Analog telephones and devices connected to the SRG always function as local telephones to the SRG. They can use the VoIP trunk to the main office using access codes or destination codes, if the VoIP trunk line pool is assigned to the device, but the main office does not have any settings or administration for these devices.

- **Access to system features**  
Analog telephones do not have a Feature key. Instead, they use a Link (\*) key to access system features. If you leave the analog telephone records at the default settings, these telephones have greater feature access on the SRG than the IP telephones in local mode. If you do not want different feature access on the analog telephones, turn the unwanted settings off as you program the telephone.

## To configure the DNs for analog devices

- 1 In Element Manager, navigate to the Active Sets panel (Configuration > Telephony > Sets > Active Sets). See the figure [Active Sets panel](#) on page 80.
- 2 Select the **Line Access** tab.

**Figure 16** Active Sets panel

The screenshot displays the 'Active Sets' configuration interface. On the left is the 'Task Navigation Panel' with a tree view where 'Active Sets' is selected. The main area is titled 'Active Sets' and has three tabs: 'Line Access', 'Capabilities and Preferences', and 'Restrictions'. The 'Line Access' tab is active, showing a table with the following data:

DN	Model	Name	Port	Pub. OLI	Priv. OLI	Fwd No Answer	Fwd Delay	Fwd Busy
233	Analog	233	0413	233	233	N/A		
234	Analog	234	0414	234	234	N/A		
235	Analog	235	0415	235	235	N/A		
236	Analog	236	0416	236	236	N/A		

Below the table are 'Copy' and 'Paste' buttons. The 'Details for DN: 235' panel is open, showing the 'Line Assignment' tab. It contains an 'Assigned Lines' table with columns: 'Line', 'Appearance Type', 'Appearances', 'Caller ID Set', 'Vmsg Set', 'Priv. Received #', and 'Pu'. There are 'Add...' and 'Delete' buttons at the bottom of this panel.

- 3 Identify the DNs for which the Model is Analog and align the settings for each with the information in the following steps.
- 4 To support outgoing number display over the PSTN, enter the public access number for the telephone in the Public OLI field (Pub. OLI).
- 5 To assign specific PSTN lines to each telephone, add the line(s) (Details for DN panel > Line Assignment tab). You would do this if, for example, you want one user to field all customer calls when the system is in local mode.
- 6 Ensure that the Appearance Type (Details for DN panel > Line Assignment tab) is set to **Ring only**.
- 7 Assign the target line to the telephone (Details for DN panel > Line Pool Access tab).
- 8 Assign both the PSTN and VoIP trunk line pools to all telephones that are allowed to make calls over the PSTN or to the main office over the VoIP trunk.



If you want the analog telephones to emulate local mode call functionality always, assign only the PSTN line pool to the analog devices.

- 9** Select the **Capabilities and Preferences** tab.
- 10** From the Details for DNs panel, select the **Capabilities** tab.
- 11** From the Handsfree list, select **None**.
- 12** Clear the **HF Answerback** check box.
- 13** Clear the **Paging** check box.
- 14** Select the **Allow Redirect** check box if you want the user to be able to call forward to the main office or redirect lines to the main office.
- 15** From the Details for DNs panel, select the **ATA Settings** tab.
- 16** From the ATA device list, select **Telephone**.



# Chapter 8

## Troubleshooting

Potential problems, probable causes, and suggested solutions for SRG-specific configuration and operating troubles are categorized under the following topics:

- [IP telephone troubleshooting](#) on page 83
- [IP terminal details](#) on page 84
- [Probable causes for redirection failure](#) on page 85
- [Troubleshooting fallback to local mode](#) on page 86
- [IP telephones manual redirection](#) on page 86
- [Cannot clear alarms using an alarm set](#) on page 87

### IP telephone troubleshooting

The table [IP telephone troubleshooting](#) on page 83 provides issues and solutions for IP Phone troubleshooting.

**Table 17** IP telephone troubleshooting (Sheet 1 of 2)

Issue / Problem	Probable Cause / Solution
Telephone does not connect to system	If an IP telephone does not display the text <code>Connecting to server</code> within two minutes after power up, the telephone was unable to establish communications with the SRG. Double check the IP configuration of the telephone and the IP connectivity to the SRG (cables, switches, and so on).
Slow connection between the handset and the Business Communications Manager	If the connection between the IP client and the SRG is slow (ISDN, dialup modem), change the preferred codec for the telephone from G.711 to G.729.
Block individual IP sets from dialing outside the system.	If you want to block one or more IP telephones from calling outside the system, use Restriction filters and assign them to the telephones you want to block. Restriction filters are set up under Configuration > Telephony > Call Security > Restriction Filters.

**Table 17** IP telephone troubleshooting (Sheet 2 of 2)

Issue / Problem	Probable Cause / Solution
One-way or no speech paths	<p>Signaling between the IP telephones and the SRG uses UDP port 7300. Voice packets are exchanged using the default RTP:</p> <p>Source port (output filters)/Destination port (input filters): 28000 through 28511 for the VoIP gateway. Output filter Destination IP is set to ALL. Input filter Destination IP is the IP address of the SRG local gateway.</p> <p>Source port (output filters)/Destination port (input filters): 5200 - 5201 for the IP telephones. Output Destination IP is set to ALL. Input filter Destination IPs are the IP range for all IP telephones (behind the firewall) in Normal mode.</p> <p>Source port (output filters)/Destination port (input filters): 51000 - 51184 for the local mode IP sets. Destination port (output filter) and Source Port (input filter) are set to ALL. Output Destination IP is set to ALL. Input filter Destination IPs are the IP range for all IP telephones (behind the firewall) in Local mode.</p> <p>UniSTIM signals use specific source and destination ports:</p> <p>Output filters: Source port, 5000; Destination port, 4100, 5100, 7300. Output filter Destination IP is the IP address of the main office TPS.</p> <p>Input filters: Source port, 4100, 5100, 7300; Destination port, 5000. Input filter Destination IPs are the IP range for all IP telephones (behind the firewall) in Normal mode.</p> <p>If these ports are blocked by the firewall or NAT, you will experience one-way or no-way speech paths.</p> <p><b>Firewall note:</b> If the firewall filter is set to Pass Outgoing and Block Incoming Except IP Phones, this allows only IP telephony registration traffic through, but blocks all other traffic, including H.323 calls on this interface. You must still specify an H.323 rule to allow IP call voice traffic.</p>

## IP terminal details

The table [IP Terminal Details](#) on page 84 summarizes the events that can be raised by the SRG. The events and details appear in Element Manager at Configuration > Telephony > Sets > Active Sets > IP Terminal Details.

Periodic retries may result in the same condition being detected over and over again. In these cases the SRG state machine uses flags to indicate that a given event has been logged.

**Table 18** IP Terminal Details (Sheet 1 of 2)

Details	Event Id	Severity	Call Server Type	Comments
SRG Started	57000	Warning	All	Indicates that the SRG process has started.
SRG Shutdown	57001	Warning	All	Indicates that the SRG process has shut down.
DN:XXX, Test Local Mode	57002	Warning	All	Test Feature
DN:XXX, Local Mode - Firmware is out of sync with Main Office Call Server.	57003	Warning	S1000	Indicates that IP set FW on main office has been upgraded and the required FW version is available on the SRG
DN:XXX, Local Mode - Set Firmware Upgrade in Progress	57004	Warning	S1000	The firmware required by the main office is being upgraded to the set,

**Table 18** IP Terminal Details (Sheet 2 of 2)

Details	Event Id	Severity	Call Server Type	Comments
DN:XXX, Normal Mode - Set Redirected to Main Office	57005	Warning	All	The set has been redirected to the main office.
DN:XXX, Local Mode - Redirection Pending (Set on call)	57006	Warning	All	The redirection of the set is pending as the set is on a call.
DN:XXX, Local Mode - Firmware Upgrade Pending (Set on call)	57007	Warning	S1000	The firmware upgrade to the set is pending as the set is on a call.
DN:XXX, Local Mode - Main Office Parameters Not Provisioned.	57008	Warning	All	The set is not provisioned to be redirected.
DN:XXX, Invalid ID (1) - No endpoint in Gatekeeper database	57250	Minor	S1000	Indicates configuration problem.
DN:XXX, Invalid ID (2) - ID unknown within the Call Server	57251	Minor	S1000	Indicates configuration problem.
DN:XXX, Invalid ID (3) - Endpoint in Gatekeeper database is Originating Call Server	57252	Minor	S1000	Indicates configuration problem.
DN:XXX, Local Mode - Net Connect Server Unreachable	57253	Major	S1000	Indicates either a configuration error, or a network connectivity error or the Net connect server has stopped.
DN:XXX, Local Mode - Main Office TPS Unreachable	57500	Major	All	Indicates either a configuration error, or a network connectivity error, or the MO TPS has stopped.
DN:XXX, Local Mode - Firmware is not available on the SRG	57501	Major	S1000	Indicates firmware required by the main office is not available in the SRG.
SRG terminated unexpectedly.	57750	Critical	All	Indicates that a critical error caused the SRG process to terminate.

## Probable causes for redirection failure

The IP telephone registration to the main office call server can fail due to improper configuration or lack of WAN connectivity. When a registration failure occurs, the error code and description is shown in the status field for the IP telephone in the IP Terminal Details field (see [IP terminal details](#) on page 84); the IP telephone remains registered with the SRG in local mode operation.

Definitive causes for registration failure depend on the main office call server. These causes can include:

- The main office is unreachable.
- There is no endpoint configured for the user id or branch user id / TN combination.

- The actual IP telephone set type at the SRG does not match MOTN set type at the main office.
- The user id is registered and not idle.
- The user id entry in the gatekeeper database points back to the originating node.

## Troubleshooting fallback to local mode

If the system reverts to local mode and the problem is not the WAN link to the main office, check for:

### 1 IP telephone firmware discrepancies

The SRG supports automatic firmware updates (see [CS 1000 considerations](#) on page 33). However, the possibility exists that a non-network reversion to local mode is caused when the IP telephone firmware has been updated on the main office and not on the SRG.

Check the IP Terminal Details tab (see [IP terminal details](#) on page 84) for this Status:

```
Firmware is Out of Sync with the Main Office Call Server
```

The preferred way of handling firmware upgrades is to install the patch onto the SRG first, then on the main office equipment.

When the IP telephone firmware is updated on the main office, the main office redirects all SRG IP telephones back to the SRG for a firmware upgrade. If the SRG has already been patched with the new firmware, the telephone is upgraded when it registers with the SRG. Once the telephone has the new firmware, the system automatically allows the telephone to reregister with the main office. If the correct firmware cannot be applied, for example because the SRG has not been upgraded with the new firmware, the telephone is redirected back to the main office.

### 2 Gatekeeper failure

If an IP telephone fails to establish communication with the gatekeeper when it tries to register to the main office, the telephone remains registered to the SRG and stays in local mode.

Troubleshoot the problem by checking the settings made when implementing the [CS 1000 information for the SRG](#) on page 44 and [Fallback configuration](#) on page 72. If you make changes, manually redirect the telephones ([IP telephones manual redirection](#) on page 86, below).

## IP telephones manual redirection

### To manually redirect an IP telephone to the main office

- 1 Access the Configuration > Telephony > Sets > Active Sets > IP Terminal Details.
- 2 Click the telephone listing that you want to redirect to normal mode.

- 3** Click the **Status** tab to view the **Status** field. If Status displays **Up**, the conversion was successful.
  - a** If the IP terminal does not register correctly with the main office, refer back to the IP Terminal Status tab, **Status** field and review the message to determine where the problem occurred. See [IP terminal details](#) on page 84.
  - b** If the conversion occurred correctly, perform basic telephony tests to ensure that the telephones are working as expected:
    - Make and receive calls.
    - Check feature access.
    - Check voice mail access

For specific information about making calls and using features, see the feature guides for the main office application.

## Cannot clear alarms using an alarm set

If you specify an alarm set (telephone) on the SRG through Element Manager, you cannot clear alarms using the alarm set. When the alarm set is in local mode, the soft keys are disabled and therefore alarms cannot be cleared.

The Alarm Set feature is not supported on SRG.





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# Appendix A

## Telephone features in normal and local mode

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The information provided here is designed for distribution to telephone users at the SRG.

The SRG50 supports the following:

- IP Phones 110, 1120E, and 1140E
- IP Phones 2001, 2002, 2004, 2007, and 2033
- IP Phone Key Expansion Module (KEM) – The IP Phone KEM is supported on an SRG with normal mode IP Phones. It does not function with local mode or test local mode IP Phones.
- IP Softphone 2050 v1/v2
- Mobile Voice Client (MVC) 2050
- WLAN Handsets 2210, 2211, and 2212
- WLAN Handsets 6120 and 6140
- WLAN Handsets 1210, 1220, and 1230
- Analog (500/2500 type) telephones
- Analog devices such as fax machines

### Normal mode

In normal mode, IP telephones have the feature set that is supported by the main office. User cards are not supplied with the SRG because the feature set depends on the main office applications. If required, obtain user cards from the main office for normal mode features.

Features available to analog and ISDN telephones are provided by the SRG and depend on the SRG applications. Consult the SRG system administrator for a complete description.

A quick reference list to the default SRG features for an analog telephone are provided in [ATA extension features](#) on page 95. Consult the SRG administrator to determine if these features have been changed.

### Local mode

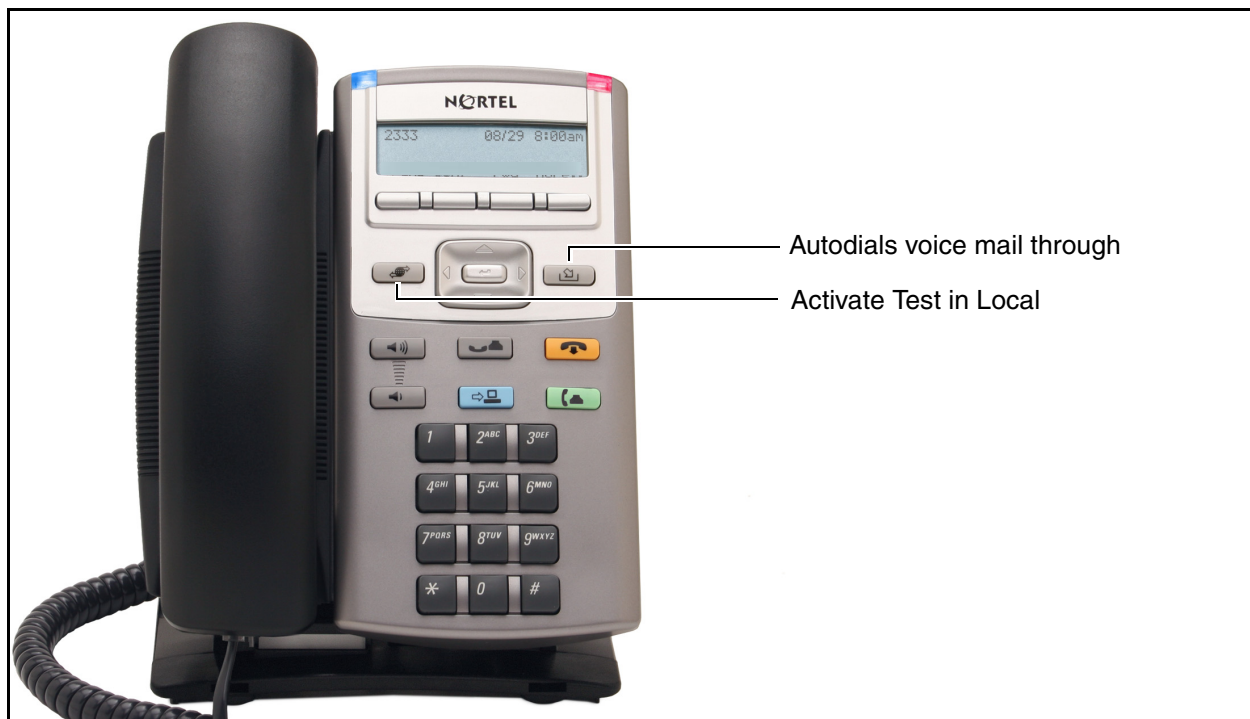
In local mode, call control and features are provided by the SRG are processed by the SRG. Access to the main office is over PSTN lines; main office telephony features and applications are not available.

If routing and destination codes are set up as suggested in [Call routing](#) on page 74, the dialing sequence for the main office is the same as in normal mode.

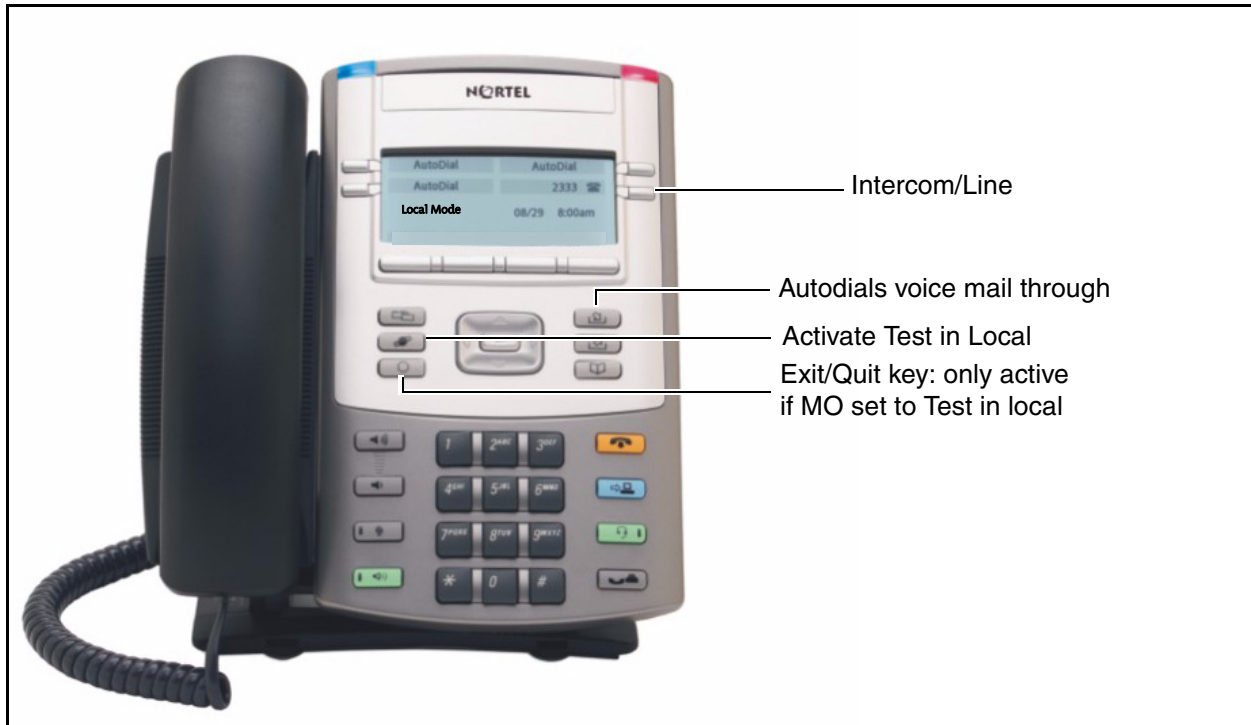
For illustrations that show the default display settings for each type of IP telephone when the phone is in local mode, refer to:

- [IP Phone 1120E in Local mode](#) on page 91
- [IP Phone 1140E in Local mode](#) on page 91
- [IP Phone 2001 in Local mode](#) on page 92
- [IP Phone 2002 in Local mode](#) on page 92
- [IP Phone 2004 in Local mode](#) on page 93
- [IP Phone 2007 in Local mode](#) on page 93
- [IP Phone 2033 in Local mode](#) on page 94
- [IP 2050 Softphone in Local Mode](#) on page 95

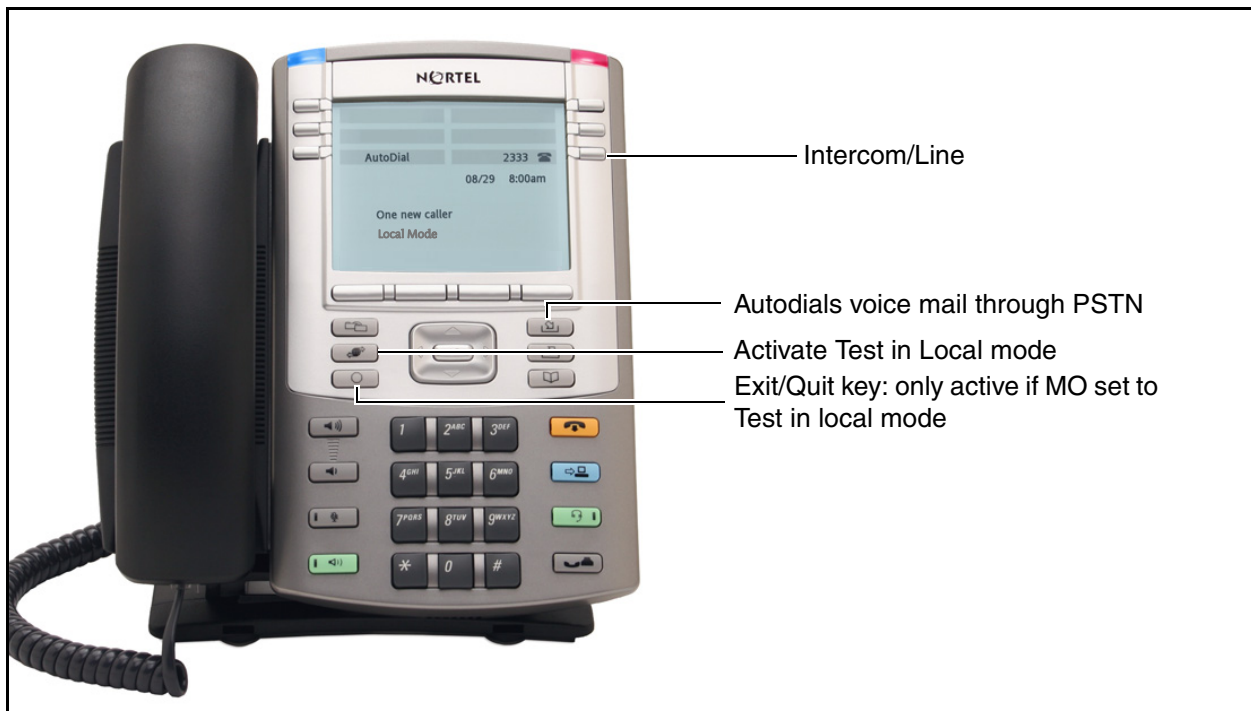
## IP Phone 1110 in Local mode



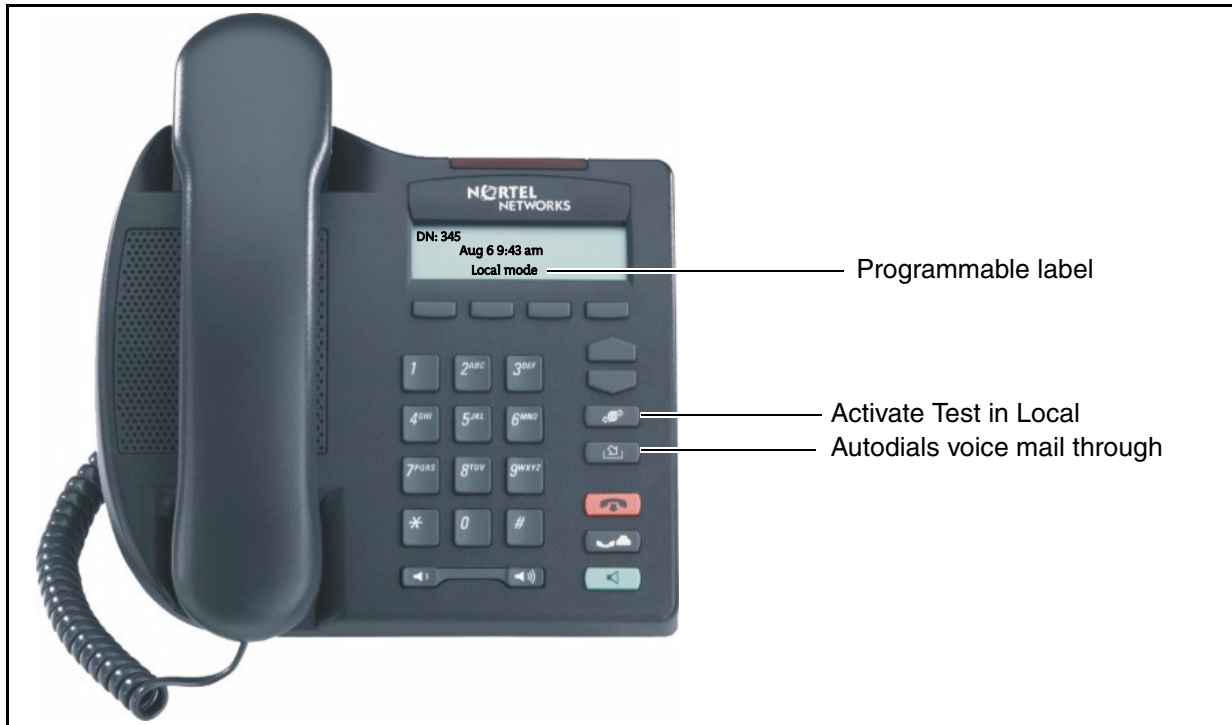
## IP Phone 1120E in Local mode



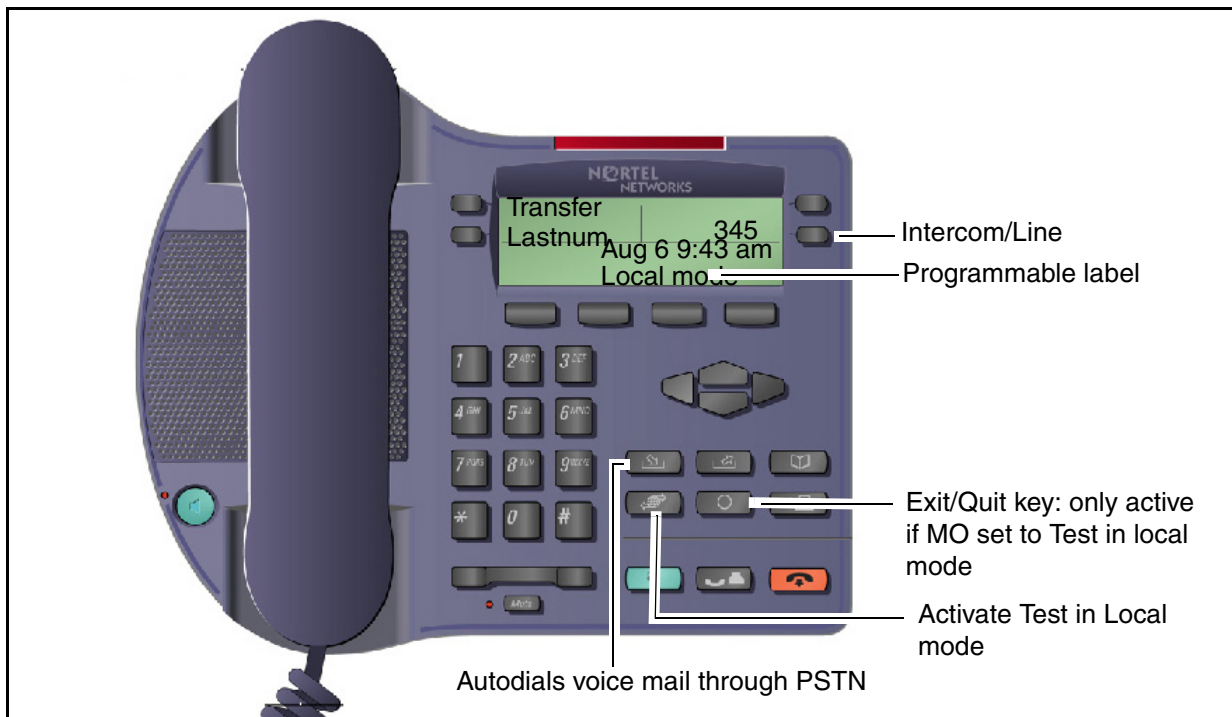
## IP Phone 1140E in Local mode



### IP Phone 2001 in Local mode



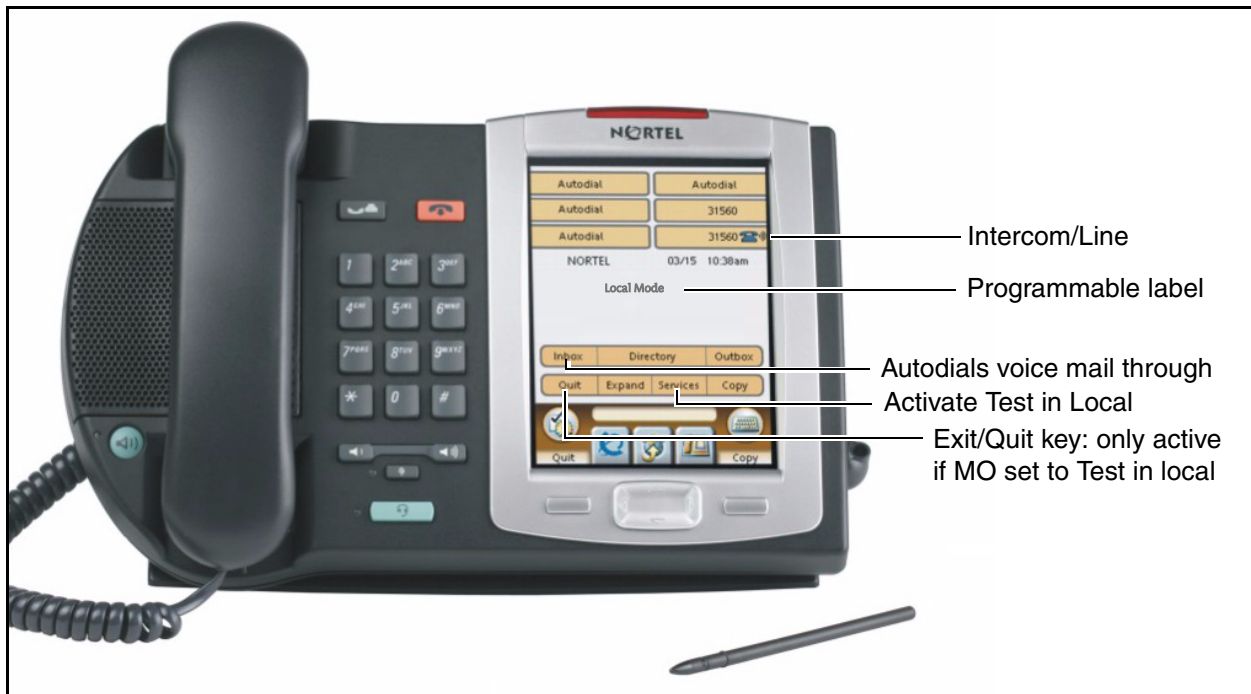
### IP Phone 2002 in Local mode



### IP Phone 2004 in Local mode



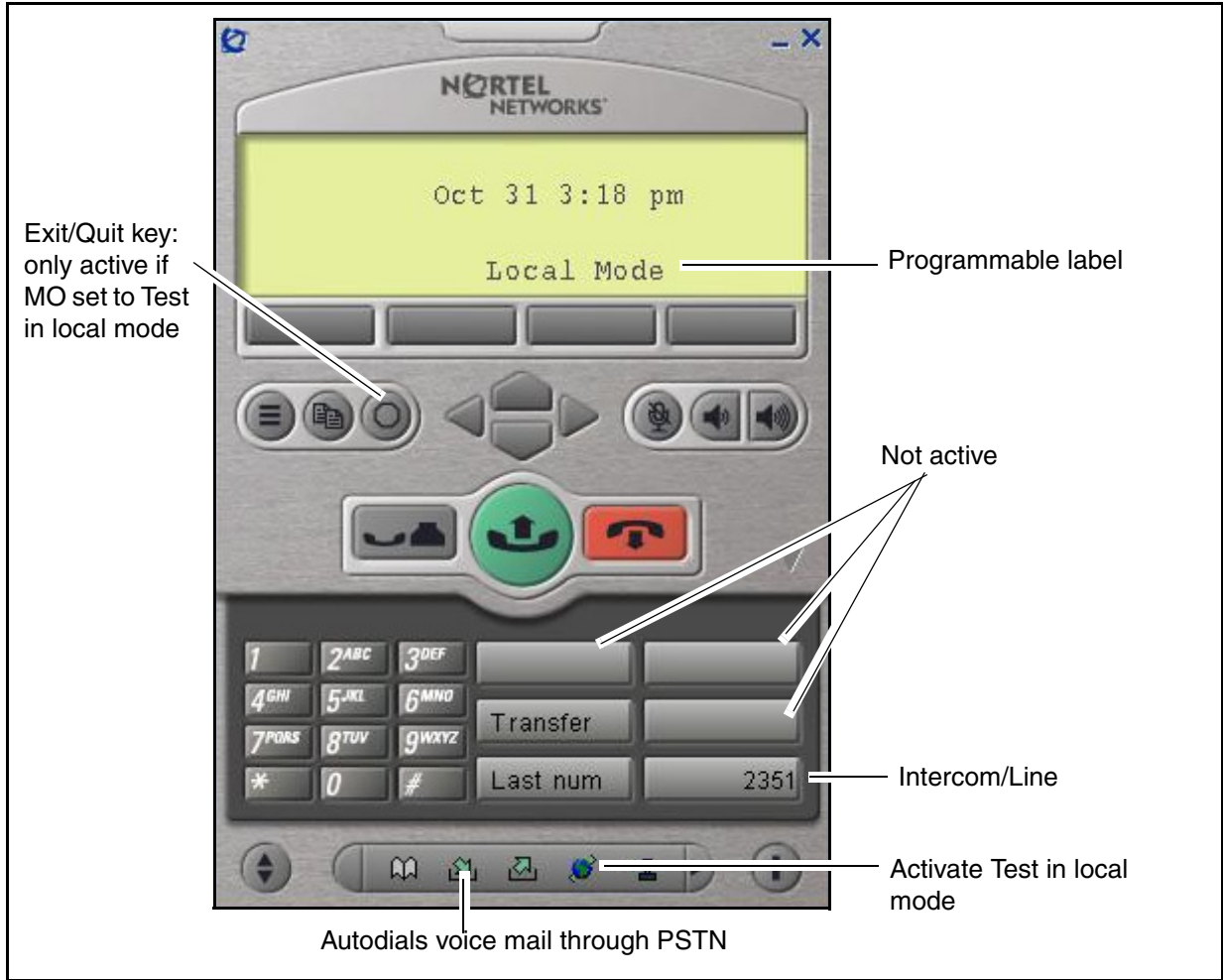
### IP Phone 2007 in Local mode



## IP Phone 2033 in Local mode



## IP 2050 Softphone in Local Mode



## ATA extension features

Analog telephones can be connected to the system through analog station modules or by installing an Analog Terminal Adapter (ATA) between the telephone and a digital station module. These telephones have only basic button configurations, so instead of using the feature key, press the link key (\*) to invoke features on the system. For specific key sequences, see the table [Link key sequences](#) on page 95.

**Table 19** Link key sequences (Sheet 1 of 2)

Feature	Activate	Cancel	Feature	Activate	Cancel
Alternate line	LINK 2		Privacy control	LINK *83	
Call Forward (local system)	LINK *4	LINK #4	Link	LINK *71	

**Table 19** Link key sequences (Sheet 2 of 2)

Feature	Activate	Cancel		Feature	Activate	Cancel
Call Forward (external system)	LINK *4 <dialled #> LINK 2	LINK #4		Pause	LINK *78	
				Timed release	LINK *72	
				Ring Again	LINK *2	LINK #2
Call parking	LINK *74			Saved Number Redial	LINK *67	
Call pick-up (Directed)	LINK *76					
Call pick-up (Group)	LINK *75			Tones	LINK *809	LINK #809
Call Queuing	LINK *801			Transfer	LINK *70	
Camp-on	LINK *82			Trunk Answer	LINK *800	
Conference call	LINK *3			Voice Call	LINK *66	
Hold Call (Exclusive)	LINK *79					
Hold Call (Public)	LINK 2			Voice messaging - Internal		
Last Number Redial	LINK *5			Access mailbox	LINK *981	
Page - Intercom	LINK *61 and zone (0 to 6)			Leave a message	LINK *980	
Page - External	LINK *62					
Page - All	LINK *63 and zone (0 to 6)					



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## Glossary

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<b>BDP</b>	Both Dialing Plans.  A dialing plan option that is supported on the main office only. The SRG supports CDP or UDP only. If the main office is running BDP, the SRG zone must be configured to allow either CDP or UDP, not both.
<b>branch office</b>	A system that is remote from the main office but provides telephony services using the main office servers. When a branch office is a survivable remote gateway, telephony services are provided by the branch office if communication with the main office is lost.
<b>call routing</b>	Coding that is configured on a system to ensure that outgoing calls are directed to the correct trunks and incoming calls are directed to the correct device(s) on the system. (see also: dialing plan)
<b>CDP</b>	Coordinated Dialing Plan.  Under the recommended Coordinated Dialing Plan, the Branch User ID can be an extension (for example, 4567). For more information about CDP, consult the main office documentation that covers dialing plans.
<b>dialing plan</b>	Each system uses a specific numbering configuration (dialing plan) that determines how calls will be handled over a private or public network. (see also: call routing)
<b>FXO</b>	Foreign eXchange Office: an interface that connects to the PSTN central office and is the interface offered on a standard telephone. Example: RJ-11 connector that allows analog connection to the central office.
<b>gatekeeper</b>	The gatekeeper is an IP network application that directs IP traffic to all the systems on the network. Parameters for both the main office and the SRG must be assigned to all gatekeepers on the network. If the gatekeeper is down, the SRG attempts to connect to the alternate gatekeeper, if there is one. If the alternate gatekeeper is also down, or there is no alternate gatekeeper, the SRG IP telephones remain registered with the main office, but calls cannot be sent to the SRG.
<b>gateway</b>	The IP portal on each system that establishes the VoIP trunk.
<b>H.323</b>	An IP gateway protocol used by both the main office and the SRG to create VoIP trunking connections.
<b>IP</b>	Internet Protocol  IP specifies the format of packets, also called datagrams, and the addressing scheme in the TCP/IP protocol suite. Where IP defines the packet and addressing scheme, Transport Control Protocol (TCP) establishes a virtual connection between a destination and a source.
<b>IP telephones</b>	Telephones that can connect directly with a TCP/IP network. Also known as internet telephones.
<b>local mode</b>	The operating mode of redirected SRG IP telephones when connectivity with the main office is unavailable.

<b>main office, main office call server</b>	The system that provides telephony services to redirected SRG IP telephones in normal mode.
<b>NCS</b>	Network Connection Server  The NCS is an H.323 gatekeeper. It provides standard H.323 gatekeeper functionality, as well as support for branch office and virtual office features.
<b>normal mode</b>	The operating mode of the SRG when connectivity with the main office is established.
<b>QoS</b>	Quality of Service  In IP telephony, QoS refers to the quality of the voice communication channel. There are inherent difficulties associated with transmitting voice over IP and quality of service is a significant challenge for service providers. QoS specifications allow service providers and their customers to establish and monitor acceptable levels of service.
<b>SIP</b>	An IP gateway protocol used by both the main office and the SRG to create VoIP trunking connections.
<b>steering codes</b>	Steering codes are similar to line pool access codes and destination codes. Steering codes determine where a call gets routed.
<b>TPS</b>	(Internet Telephone) Terminal Proxy Server  A TPS controls the connection between IP telephones.
<b>UDP</b>	Uniform Dialing Plan  Each location within the network is assigned a Location Code. On a private network, this code precedes the directory number of the telephone being dialed. Depending on routing configuration, this number may be part of the destination code, or it may automatically be dialed out when the appropriate destination code is dialed before the directory number. Under the Uniform Dialing Plan (UDP), the SRG must include this code in the BUID.
<b>UDP</b>	User Datagram Protocol  A member of the TCP/IP protocol suite that transports data as a connectionless protocol, using packet switching. Generally, ports on the SRG support UDP.
<b>VoIP trunk</b>	Voice over IP trunk  A pathway between two systems that allows voice to be transmitted over an IP connection.
<b>WAN</b>	Wide Area Network  A computer network that spans a relatively large geographical area. The largest WAN in existence is the Internet.
<b>ZDP</b>	Zone Digit Prefix  The main office appends this number to an SRG local-PSTN call dialed from an SRG IP telephone. The number differentiates the call from a main office local-PSTN call dialed by a main office telephone.

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